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Tzannes

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(54) **IMPULSE NOISE MANAGEMENT**

(75) Inventor: **Marcos C. Tzannes**, Orinda, CA (US)

(73) Assignee: **TQ Delta, LLC**, Austin, TX (US)

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H04L 5/16 (2006.01)

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USPC **375/219**; 375/224; 375/284; 375/346

(58) **Field of Classification Search**
USPC 375/219, 224, 285, 346; 714/774, 714/786
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

3,699,457 A 10/1972 Wright
4,845,466 A 7/1989 Hariton et al.
5,537,437 A 7/1996 Kaku et al.
5,794,136 A 8/1998 Buchwald et al.
6,498,808 B1 12/2002 Tzannes
6,647,070 B1 11/2003 Shalvi et al.
6,654,410 B2 11/2003 Tzannes
6,683,855 B1 1/2004 Bordogna et al.
6,788,923 B2 9/2004 Tsujishita et al.
6,920,194 B2 7/2005 Stopler et al.
7,050,419 B2 5/2006 Azenkot et al.
7,162,676 B2 1/2007 Coleman et al.
7,428,669 B2 9/2008 Cioffi
7,673,221 B2 3/2010 Li et al.
2002/0069038 A1 6/2002 Cooper
2002/0150167 A1 * 10/2002 Demjanenko et al. 375/259
2006/0078044 A1 4/2006 Norrell et al.
2008/0232444 A1 9/2008 Tzannes
2009/0086800 A1 4/2009 Cioffi
2010/0229075 A1 * 9/2010 Yousef et al. 714/784

OTHER PUBLICATIONS

ITU-T, G.992.3, Series G: Transmission Systems and Media, Digital Systems and Networks, International Telecommunication Union, Jul. 2002.

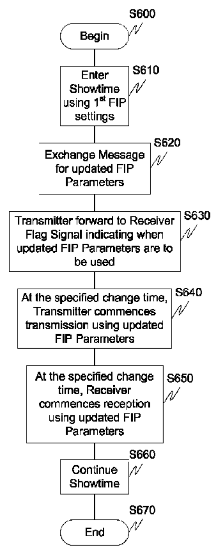
(Continued)

Primary Examiner — Jean B Corrielus
(74) *Attorney, Agent, or Firm* — Jason H. Vick; Sheridan Ross, PC

(57) **ABSTRACT**

Evaluation of the impact of impulse noise on a communication system can be utilized to determine how the system should be configured to adapt to impulse noise events. Moreover, the system allows for information regarding impulse noise events, such as length of the event, repetition period of the event and timing of the event, to be collected and forwarded to a destination. The adaptation can be performed during one or more of Showtime and initialization, and can be initiated and determined at either one or more of a transmitter and a receiver.

32 Claims, 7 Drawing Sheets



OTHER PUBLICATIONS

Itu-T, G.993.1, Series G: Transmission Systems and Media, Digital Systems and Networks, International Telecommunication Union, Revision 2004.

Proposed revisions to G.992.4 and G.992.3&5 regarding impulse noise protection, ITU-Telecommunication Standardization Sector, SBC, BellSouth, BT. Singapore, Jan. 19-Jan. 24, 2004.

International Search Report for International (PCT) Patent Application No. PCT/US2005/006842, mailed Oct. 5, 2005.

Written Opinion for International (PCT) Patent Application No. PCT/US2005/006842, mailed Oct. 5, 2005.

International Preliminary Report on Patentability for International (PCT) Patent Application No. PCT/US2005/006842, dated Sep. 5, 2006.

First Office Action for Chinese Patent Application No. 200580006738.8, mailed Oct. 23, 2009.

Restriction Requirement for U.S. Appl. No. 10/597,482, mailed May 7, 2009.

Official Action for U.S. Appl. No. 10/597,482, mailed Jun. 30, 2009.

Official Action for U.S. Appl. No. 10/597,482, mailed Dec. 2, 2009.

Official Action for U.S. Appl. No. 10/597,482, mailed Jul. 19, 2010.

Official Action for U.S. Appl. No. 10/597,482, mailed Feb. 9, 2011.

Official Action for Indian Patent Application No. 2155/KOLNP/2006, dated Feb. 18, 2011.

Moulin et al., "Discrete-multitone-based ADSL and VDSL systems performance analysis in an impulse noise environment—Advances in Communication Cables Technology," IEE Proceedings, Nov. 3, 2003, vol. 150(6), pp. 273-278.

Toumpakaris et al., "A Simple Byte-Erasure Method for Improved Impulse Immunity in DSL," 2003 IEEE International Conference on Communications, May 2003, vol. 1 of 5, pp. 2426-2430.

Official Communication for European Patent Application No. 05724394.1, dated Aug. 19, 2010.

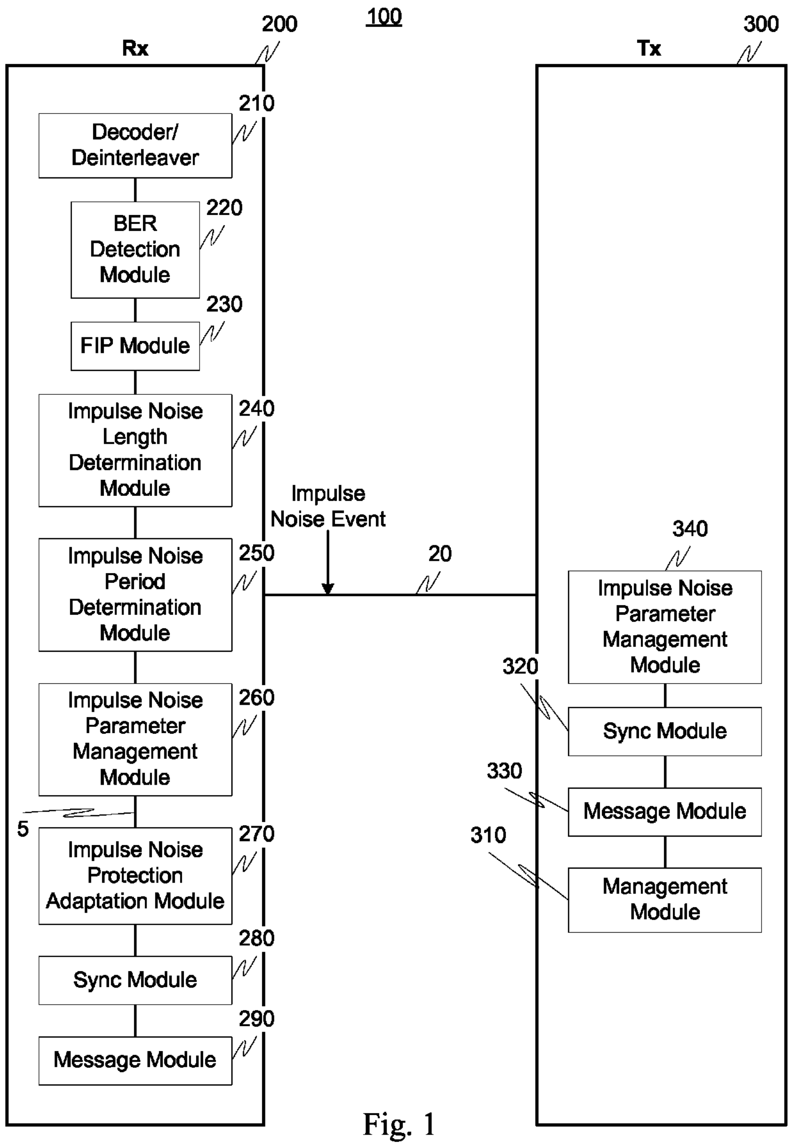
Extended European Search Report for European Patent Application No. 10006702.4, dated Aug. 13, 2010.

Second Office Action (including translation) for Chinese Patent Application No. 200580006738.8, issued Jun. 30, 2011.

Examiner's Report for Canadian Patent Application No. 2,555,757, mailed Oct. 17, 2012.

Invitation to Pay Additional Fees for International (PCT) Patent Application No. PCT/US2005/006842, mailed Jul. 28, 2005.

* cited by examiner



100
200
Rx
300
Tx
210
Decoder/ Deinterleaver
220
BER Detection Module
230
FIP Module
240
Impulse Noise Length Determination Module
Impulse Noise Event
250
Impulse Noise Period Determination Module
20
340
Impulse Noise Parameter Management Module
320
Sync Module
260
Impulse Noise Parameter Management Module
330
Message Module
5
270
Impulse Noise Protection Adaptation Module
310
Management Module
280
Sync Module
290
Message Module

Fig. 1

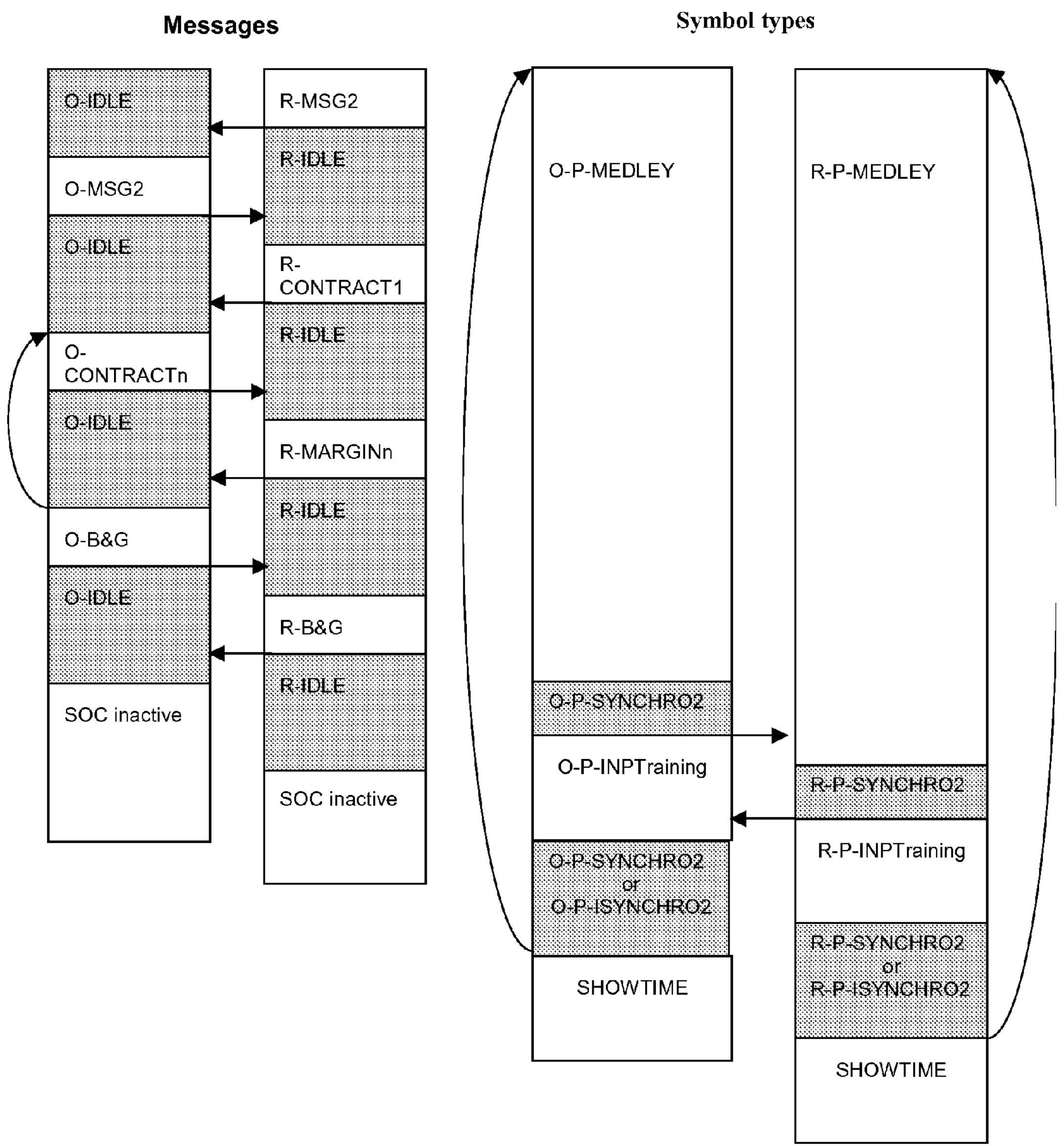

Messages
Symbol types
O-IDLE
O-MSG2
O-IDLE
O-
CONTRACTn
O-IDLE
O-B&G
O-IDLE
SOC inactive
R-MSG2
R-IDLE
R-
CONTRACT1
R-IDLE
R-MARGINn
R-IDLE
R-B&G
R-IDLE
SOC inactive
O-P-MEDLEY
O-P-SYNCHRO2
O-P-INPTraining
O-P-SYNCHRO2
or
O-P-ISYNCHRO2
SHOWTIME
R-P-MEDLEY
R-P-SYNCHRO2
R-P-INPTraining
R-P-SYNCHRO2
or
R-P-ISYNCHRO2
SHOWTIME

Fig. 2

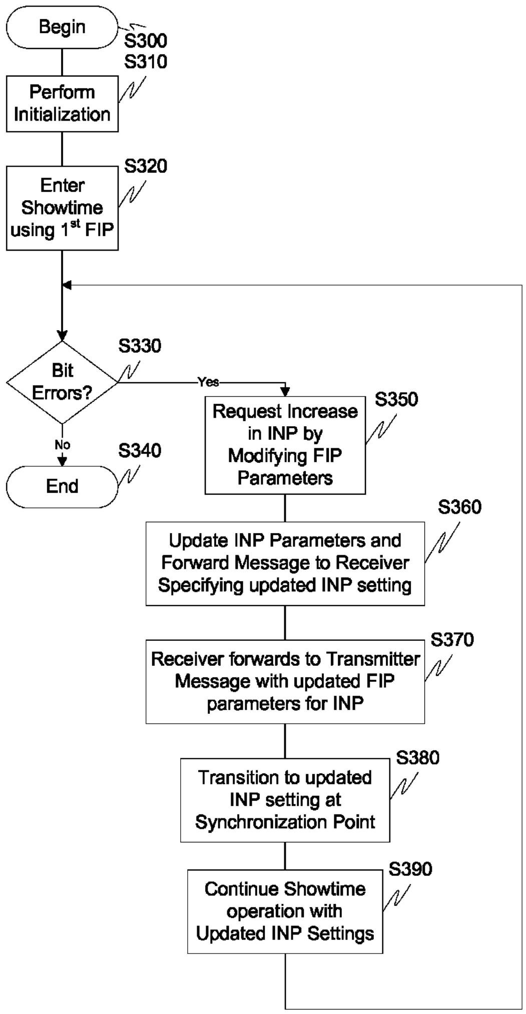

Begin
S300
S310
Perform Initialization
S320
Enter Showtime using 1st FIP
S330
Bit Errors?
Yes
No
S340
End
S350
Request Increase in INP by Modifying FIP Parameters
S360
Update INP Parameters and Forward Message to Receiver Specifying updated INP setting
S370
Receiver forwards to Transmitter Message with updated FIP parameters for INP
S380
Transition to updated INP setting at Synchronization Point
S390
Continue Showtime operation with Updated INP Settings

Fig. 3

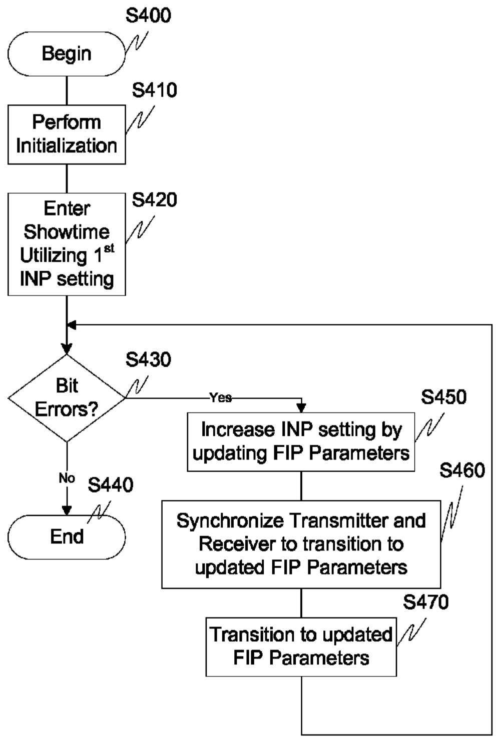

S400
Begin
S410
Perform Initialization
S420
Enter Showtime Utilizing 1st INP setting
S430
Bit Errors?
Yes
No
S440
End
S450
Increase INP setting by updating FIP Parameters
S460
Synchronize Transmitter and Receiver to transition to updated FIP Parameters
S470
Transition to updated FIP Parameters

Fig. 4

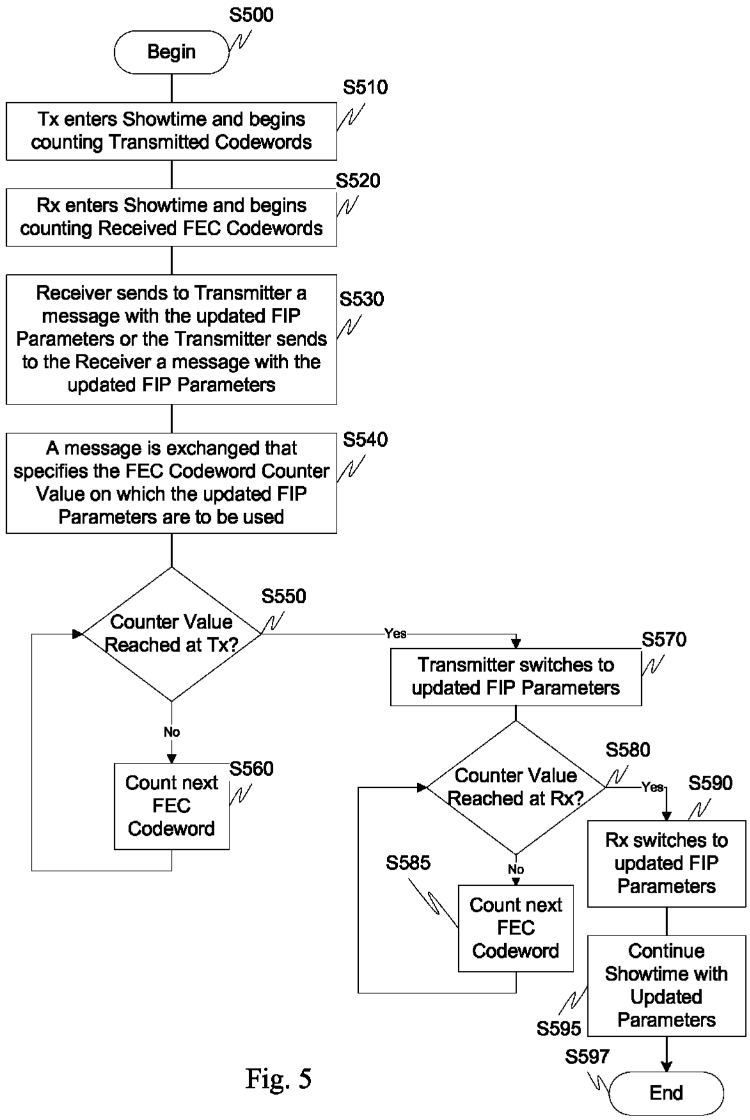

S500
Begin
S510
Tx enters Showtime and begins counting Transmitted Codewords
S520
Rx enters Showtime and begins counting Received FEC Codewords
S530
Receiver sends to Transmitter a message with the updated FIP Parameters or the Transmitter sends to the Receiver a message with the updated FIP Parameters
S540
A message is exchanged that specifies the FEC Codeword Counter Value on which the updated FIP Parameters are to be used
S550
Counter Value Reached at Tx?
Yes
No
S560
Count next FEC Codeword
S570
Transmitter switches to updated FIP Parameters
S580
Counter Value Reached at Rx?
Yes
No
S585
Count next FEC Codeword
S590
Rx switches to updated FIP Parameters
Continue Showtime with Updated Parameters
S595
S597
End

Fig. 5

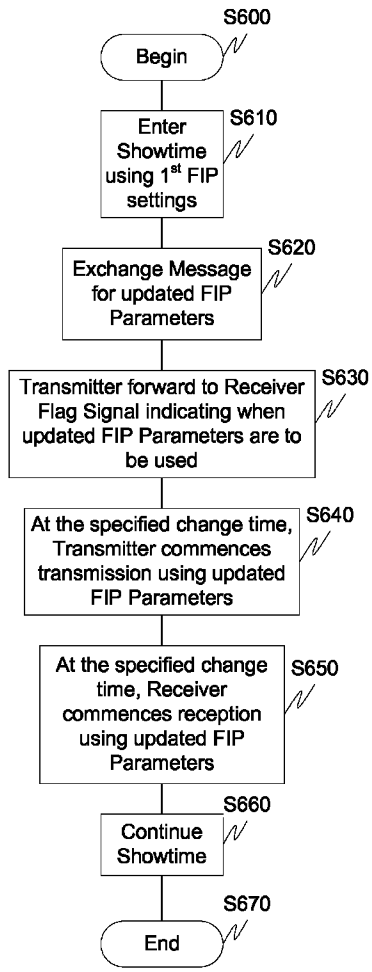

S600
Begin
Enter Showtime using 1st FIP settings
S610
Exchange Message for updated FIP Parameters
S620
Transmitter forward to Receiver Flag Signal indicating when updated FIP Parameters are to be used
S630
At the specified change time, Transmitter commences transmission using updated FIP Parameters
S640
At the specified change time, Receiver commences reception using updated FIP Parameters
S650
Continue Showtime
S660
End
S670

Fig. 6

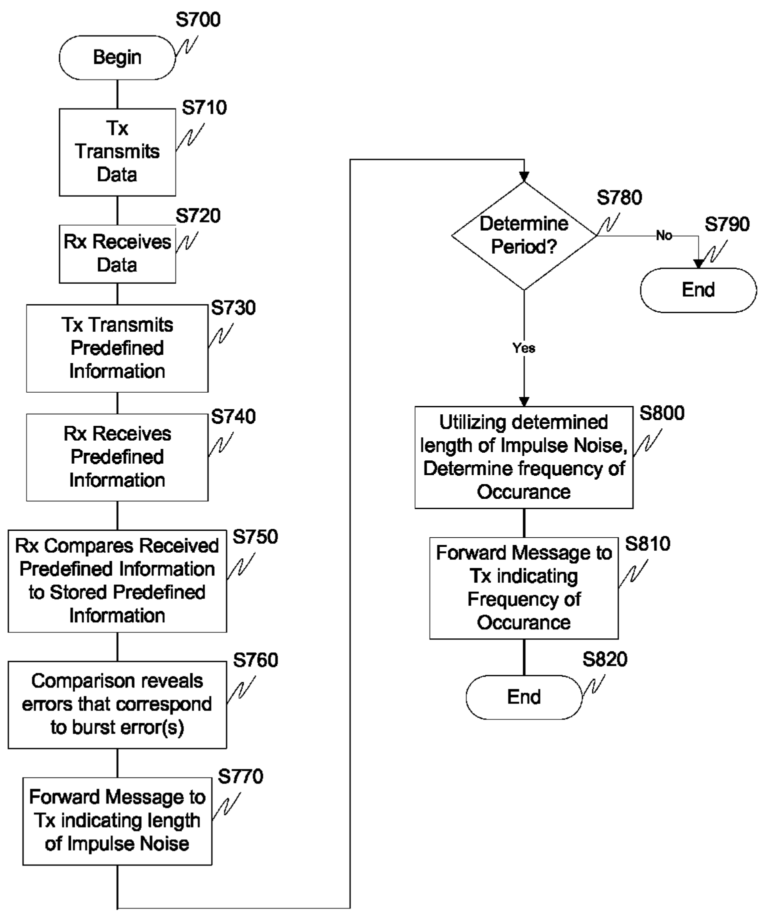

S700
Begin
S710
Tx Transmits Data
S720
Rx Receives Data
S730
Tx Transmits Predefined Information
S740
Rx Receives Predefined Information
S750
Rx Compares Received Predefined Information to Stored Predefined Information
S760
Comparison reveals errors that correspond to burst error(s)
S770
Forward Message to Tx indicating length of Impulse Noise
S780
Determine Period?
No
S790
End
Yes
S800
Utilizing determined length of Impulse Noise, Determine frequency of Occurance
S810
Forward Message to Tx indicating Frequency of Occurance
S820
End

Fig. 7

IMPULSE NOISE MANAGEMENT

RELATED APPLICATION DATA

This application is a continuation of U.S. application Ser. No. 10/597,482, filed Jul. 27, 2006 now abandoned, which is a national stage application under 35 U.S.C. 371 of PCT Application No. PCT/US2005/006842, filed Mar. 3, 2005, which claims the benefit of and priority under 35 U.S.C. §119(e) to U.S. Provisional Application No. 60/549,804, entitled "On-Line Impulse Noise Protection (INP) Adaptation," filed Mar. 3, 2004, and U.S. Provisional Application No. 60/555,982, entitled "Impulse Noise Protection (INP) Training," filed Mar. 24, 2004, each of which are incorporated herein by reference in their entirety.

BACKGROUND

1. Field of the Invention

This invention generally relates to communication systems. In particular, an exemplary aspect of this invention relates to impulse noise protection adaptation. Another exemplary aspect of this invention relates to impulse noise length and period determination and use thereof for impulse noise protection adaptation.

2. Description of Related Art

Communications systems often operate in environments that produce impulse noise. Impulse noise is a short-term burst of noise that is higher than the normal noise that typically exists in a communication channel. For example, DSL systems operate on telephone lines and experience impulse noise from many external sources including telephones, AM radio, HAM radio, other DSL services on the same line or in the same bundle, other equipment in the home, etc. It is standard practice for communications systems to use interleaving in combination with Forward Error Correction (FEC) to correct the errors caused by impulse noise. Standard initialization procedures in ADSL and VDSL systems are designed to optimize performance (data rate/reach) in the presence "stationary" crosstalk or noise. Impulse noise protection is handled with interleaving and FEC, but the current xDSL procedure at least does not provide specific states to enable training for the selection of the appropriate interleaving and FEC parameters.

An exemplary problem associated with traditional communication systems is that they use traditional Signal to Noise Ratio (SNR) measurement techniques to determine the SNR of the channel. These traditional techniques assume that the noise is stationary and does not contain non-stationary components such as impulse noise. The most common method for measuring the SNR is to calculate the mean-square error of the received signal based on a known transmitted signal, which is described in the ADSL series of ITU G.992.x standards and the VDSL series of ITU G.993.x standards, which are incorporated herein by reference in their entirety. These traditional methods for measuring SNR do not correctly measure the impact of impulse noise and do not have the noise capability to determine how the system should be configured to handle impulse noise.

There has been proposed that there is a need in ADSL and VDSL systems to provide robust error-free performance in the presence of high, real-world impulse noise. A specific proposal recommends that the standard impulse noise protection (INP) values are extended to values of 4, 8, 16 and 32 in order to handle high levels of impulse noise. Impulse noise protection is defined in the ADSL2 Standard G.992.3, which is incorporated herein by reference in its entirety, as the number of impulse noise corrupted DMT symbols that can be corrected by the FEC and interleaving configuration. Specifically, G.992.3 defines the following variables:

$$INP=1/2*(S*D)*R/N$$

$$S=8*N/L$$

$$\text{Latency (or delay)}=S*D/4$$

$$\text{Line Rate (in kbps)}=L*4$$

where N is the codeword size in bytes, R is the number of parity (or redundancy) bytes in a codeword, D is the interleaver depth in number of codewords, and L is the number of bits in a DMT symbol.

If K is the number of information bytes in a codeword then:

$$N=K+R$$

and the user data rate is approximately equal to:

$$L*4*K/N.$$

In general, DSL systems (such as the one defined in ADSL G.992.x or VDSL G.993.x) use the FEC and Interleaving Parameters (FIP) characterized by the set of parameters (N, K, R, D). Using these parameters, the Burst Error Correction Capability (BECC) in bytes can be simply calculated as:

$$BECC=D*R/2\text{ bytes}$$

where BECC is defined as the number of consecutive byte errors that can be corrected by the receiver. Note that if the receiver uses more intelligent decoding schemes, such as erasure detection, it is possible to correct even more than D*R/2 bytes. It also follows from above that INP=BECC/L.

The proposal further recommends that the higher INP values are achieved by increasing the amount of FEC redundancy while keeping the same system latency and the same interleaver memory at the expense of user data rate or excess margin. Since, on phone lines without excess margin, there is clearly a trade-off between high impulse noise protection values and user data rate, it would be advantageous to try to maximize the user data rate by finding the minimum impulse noise protection value that can provide adequate impulse noise protection. The current technique includes the steps of an operator, or service provider, configuring the ADSL connection with a specific noise protection value, the ADSL connection is initialized and the transceivers enter into steady state data transmission (i.e., Showtime), and if the connection is stable, i.e., error-free, then the service is acceptable and the process ends. If there are bit errors, then the process is repeated with the operator, or service provider, configuring the ADSL connection with another specific INP value.

One exemplary problem with this approach is that it is time consuming and can result in sub-optimum user rates. This is illustrated with reference to the following examples:

Example 1: Assume that for a particular DSL connection there is high impulse noise and the required INP is 8. As a result, if the service provider uses a first INP configuration of 2, the DSL connection will not be error free. Therefore, the service provider needs to configure a higher INP value and reinitialize the connection. If a value of 4 is used as a second INP value, it still will not provide adequate impulse noise protection and bit errors will occur. Again, the service provider will need to configure a higher INP value until the correct value of 8 is configured. Clearly, the connection needs to be re-initialized every time there is a new INP configuration chosen and this trial and error technique proves to be very time consuming

Example 2: Assume that for a particular DSL connection there is high-impulse noise and the required INP is 4. As a result, if the service provider uses a first INP configuration of 2, the connection will not be error free. Therefore, the service provider needs to configure a higher INP value and reinitialize the connection. In order to save time and not go through the number of initializations has occurred in Example 1, the service provider simply configures the system to the maximum INP value of 32. Obviously, there will be no bit errors with INP=32 since this connection needs only an INP value of 4. As a result, user data throughput is greatly degraded since the additional FEC redundancy will be three times higher than what is actually needed. For example, if the INP of 4 requires 10 percent FEC redundancy, an INP of 32 requires 40 percent FEC redundancy which results in a 30 percent decrease in user data rate.

SUMMARY

In additional to the above drawbacks, the related systems do not have the ability to actually measure the length or repetition period of impulse noise which can both be used to determine an appropriate impulse noise protection setting.

Exemplary aspects of this invention relate to determining the impact of impulse noise on a communication system and the capability to determine how the system should be configured to handle the impulse noise event.

An exemplary aspect of this invention determines the impact of impulse noise by transmitting and receiving using a plurality of different FEC and interleaving parameter settings. For each FEC and Interleaving Parameter (FIP) setting, the received signal quality is determined by, for example, detecting if there are bit errors after the receiver performs the FEC decoding and deinterleaving. Based on this, the appropriate FIP setting is selected and used for transmission and reception.

A plurality of FIP settings can be used for transmission and reception. In accordance with one particular aspect of this invention, the system can transition from one FIP setting to another FIP setting without going through the startup initialization procedure such as the startup initialization sequence utilized in traditional xDSL systems. For example, an xDSL system that implements the systems and methods described herein could start using an FIP setting of (N=255, K=247, R=8, D=64) and then transition to an FIP setting of (N=255, K=239, R=16, D=64) without re-executing the startup initialization procedure.

Knowing that the first FIP setting has a BECC=256 bytes and the second setting has a BECC=512 bytes, means, for example, that the second setting can correct an impulse that causes twice as many bit errors as the first FIP setting. On the other hand, the first FIP setting has less FEC parity (overhead) which results in a higher information (net) data rate for the user during Showtime. This is also illustrated by the fact that K, the number of information bytes per code word, is higher for the first FIP setting. For each of the FIP settings, the receiver detects whether there are bit errors after the decoding and deinterleaving process. This detection can be done by, for example, by performing a cyclic redundancy check (CRC) after the decoding and deinterleaving process is complete as defined in the ITU standard G.992.x. In general, the CRC is a well-known method for detecting bit errors. Since impulse noise occurs at random times, this system can operate using a particular FIP setting for a period of time that is sufficient to encounter the impulse noise. In the simple example illustrated above, only the K and R values were modified. It should be appreciated however, that the systems and methods of this invention are not limited thereto but rather can be extended to include the modification of any one or more FIP parameter(s).

The process of determining the impact of impulse noise by transmitting and receiving using a plurality of FIP settings can be done while in steady-state transmission, i.e., Showtime for DSL systems, when user information bits are being transmitted.

The process of determining the impact of impulse noise by transmitting and receiving using a plurality of FIP settings, can be done during a special impulse noise training period during which the system is not actually transmitting user data. In accordance with this exemplary aspect of the invention, the standard xDSL procedure is modified to include the capability of measuring the effectiveness of a chosen impulse noise protection (INP) setting during initialization and having receiver-controlled updates of transmission parameters that control the INP setting, e.g., FEC parameters and interleaving parameters, during initialization.

A new initialization state is included in the xDSL initialization procedure that provides the capability to measure the effectiveness of the current INP settings. The new initialization state will be referred to as the INPTraining state and it can, for example, follow the exchange of the Showtime transmission parameters such as the bi/gi table, trellis coding, tone reordering, FEC/interleaving parameters, etc. It is important to note that steady-state transmission during which user information is transmitted is known as "Showtime" in XDSL systems and that standard ITU G.992.3 ADSL systems and ITU VDSL G.993.3 systems include an exchange phase in initialization during which the Showtime parameters are exchanged, see, for example, G.992.3 and G.993.1.

During this INPTraining state, the transceivers transmit and receive using at least one of the standard Showtime functions using the Showtime parameters, e.g., bi/gi table, FEC/interleaving parameters, etc., that are exchanged during the previous exchange phase. These functions include at least one of Showtime PMD functions, e.g., bi/gi table, trellis coding, tone reordering, etc., PMS-TC functions, such as framing and FEC/interleaving, and TPS-TC functions. During the INPTraining state, the TPS-TC can transmit idle ATM cells, HDLC flags, or 64/65 idle packets depending on the TPS-TC type.

At the receiver, the CRC, the FEC, the TPS-TC error detection capabilities, and other receiver functions can be used to determine whether the INP setting is adequate for the current impulse noise conditions on the line. The receiver can also use these receiver functions to automatically and dynamically determine what the correct INP setting should be. If the current INP setting is not adequate, a new set of Showtime transmission parameters can be exchanged and the process repeated by reentering the INPTraining state using the newly exchanged Showtime transmission parameters. If, on the other hand, the current INP setting is adequate, the transceivers can enter into Showtime.

An exemplary advantage associated with this aspect of the invention is that the receiver can measure the effectiveness of the current INP setting and can make updates to the INP setting based on these measurements during initialization. This is important because the receiver generally has the most knowledge of channel conditions, receiver functionality, and processing capability. In general, the receiver has the best capability to make the necessary trade offs between data rate, latency, excess margin, FEC redundancy, coding gain, and the like. Current related ADSL systems have the operator servicing lines that are experiencing high impulse noise by trying various INP settings in an attempt to find an error-free operating mode. However, since the operator is not capable of

making the receiver trade offs stated above, such as the data rate, latency, and the like, the process will often lead to a sub-optimum result.

Another exemplary aspect of this invention relates to determining the length and/or repetition period of impulse noise events in order to select an appropriate INP setting.

For example, the transceivers can transmit and receive using at least one of the standard Showtime functions and parameters such as the bi/gi table, FEC/interleaving parameters, and the like. These functions include the Showtime PMD functions, such as the bi/gi table, trellis coding, tone reordering and the like, PMS-TC functions (such as framing and FEC interleaving) and TPS-TC functions. The TPS-TC may transmit ATM cells, HDLC packets, or 64/65 packets depending on, for example, the TPS-TC type.

At the receiver, the CRC, the FEC, the TPS-TC error detection capabilities, and other receiver functions can be used to determine the length and the period of the impulse noise and whether an INP setting is adequate for the current impulse noise conditions on the line.

For example, the receiver can determine the length of an impulse noise event. The length of impulse noise events can be determined in a number of ways. For example, they can be determined as:

1. The length of the impulse in time, e.g., how many microseconds the impulse power is above a specific noise level (e.g. the above the stationary channel noise)
2. The number received bits that are affected by the impulse noise, e.g. how many received bits are in error in a specific sliding time window or how many consecutive bits are in error in a stream
3. The number of received ATM cells that are affected by the impulse noise, e.g. how many ATM cells contain bits that are in error. This may be detected using a standard ATM HEC, which is a CRC that covers the ATM header bits. Alternatively this may detected by checking the ATM payload bits by, for example, transmitting a predefined bit pattern that is known by the receiver
4. The number of 64/65 packets that are affected by the impulse noise, e.g. how many packets contain bits that are in error (the 64/65 CRC can be used for this)
5. The number of received DMT symbols that are affected by the impulse noise, e.g., the measured noise in a DMT symbol is above a predefined threshold which results in most (if not all) the bits in that DMT symbol being in error
6. The number of FEC codewords that are affected by the impulse noise, e.g., how many codewords have an uncorrectable number of bit errors, i.e., the number of bit errors in a codeword exceeds the number of bit that are correctable by the FEC code

In accordance with one exemplary aspect, the FEC correction capability and interleaving is turned off when trying to determine the length of the impulse noise. For example, the FEC may be configured so that there are no parity bits (i.e., R=0) or the FEC may be disabled altogether (i.e., no codewords are sent). Additionally, for example, the interleaving may be disabled by setting the interleaver depth to 1 (i.e., D=1). Disabling the FEC and interleaving is beneficial when trying to determine the length of the impulse noise based on affected bits, affected code words and/or affected packets. This is true since when the FEC/interleaving is enabled, the impulse noise event will be spread over a large time period and it will be more difficult to determine the length of the original impulse.

The exemplary techniques discussed herein that are used to determine the length of the impulse event can also be extended to determining the repetition period of the impulse noise event, i.e., how often is impulse noise occurring. The repetition period can be important, because the period has an effect on the FIP setting that is used. For example, if the interleaving spreads an impulse noise event over a period of time that exceeds the impulse period, then the interleaver could combine multiple impulse noise events together. As a result, the FEC correction capability may have to be increased. In order to determine the repetition period of an impulse noise event, the receiver can first detect the impulse noise event as discussed above and then determine how often the impulse noise events occur. For example, periodic impulse noise due to AC power lines occurs at a 120 Hz reception rate, or approximately every 8 ms.

In the case where the impulse noise does not have a fixed length, i.e., where the impulse noise varies over time, the receiver can attempt to determine the maximum impulse noise length. This involves measuring several impulse noise events and determining the impulse noise event with the maximum length. In practice, it is likely that the impulse noise will have varying length and it is important that the FEC/interleaving settings be configured to handle the maximum, e.g., worst case, impulse noise event.

Once the receiver determines the (maximum) length of the impulse noise and/or the repetition period of the impulse noise, this information can be sent to the transmitting modem. In particular, when the receiving modem, such as a Customer-Premises (CPE) modem determines the (maximum) length of the impulse and/or the repetition period of the impulse, the CPE modem could send the information to a Central Office (CO) modem in a message. The length of the impulse length event can be defined and specified in the message in terms of time, received bits, ATM cells, 64/65 packets, DMT symbols, or the like. The CO modem could then, for example, provide this information to the CO-MIB, which is the management interface that is used by the operator or service provider to configure the modems. For example, based on this information, the operator may configure the modems to a different INP value, data rate, latency, or the like. This process could also be automated such that the message received by the CO modem allows automatic reconfiguration to adjust, for example, INP values, data rate, latency, or the like.

The exemplary aspects of the invention that are used to determine the length and/or repetition period of the impulse noise event can be performed during initialization and/or Showtime. During initialization, the method can perform during an INPTraining state such as the one described herein. During Showtime the method can be performed, for example, during a Showtime INP adaptation phase as described herein.

These and other features and advantages of this invention are described in, or are apparent from, the following description of the embodiments.

BRIEF DESCRIPTION OF THE DRAWINGS

The features of this invention will be described in relation to the figures, wherein:

FIG. **1** is a functional block diagram illustrating an exemplary impulse noise adaptation system according to this invention;

FIG. **2** illustrates an exemplary initialization state machine that includes an INPTraining state according to this invention;

FIG. **3** is a flowchart outlining an exemplary method for impulse noise protection adaptation during Showtime according to this invention;

FIG. **4** is a flowchart illustrating an exemplary receiver optimized method for impulse noise protection adaptation according to this invention;

FIG. **5** is a flowchart illustrating an exemplary method for forward error correction synchronization according to this invention;

FIG. **6** is a flowchart illustrating an exemplary method for flag signal synchronization according to this invention; and

FIG. **7** is a flowchart illustrating an exemplary method for determining impulse noise length and period according to this invention.

DETAILED DESCRIPTION

The exemplary embodiments of this invention will be described in relation to adapting impulse noise parameters as well as measuring impulse noise length and repetition period within an xDSL environment. However, it should be appreciated that, in general, the systems and methods of this invention can be applied and will work equally well with any type of communication system in any environment.

The exemplary systems and methods of this invention will be described in relation to xDSL modems and associated communication hardware, software, and communication channels. However, to avoid unnecessarily obscuring the present invention, the following description omits well-known structures and devices that may be shown in block diagram form or otherwise summarized.

For purposes of explanation, numerous details are set forth in order to provide a thorough understanding of the present invention. It should be appreciated however, that the present invention may be practiced in a variety of ways beyond the specific details set forth herein. For example, the systems and methods of this invention can generally be applied to any type of system within any environment in which the impulse noise length and/or period is desired, or for which impulse noise adaptation is desired.

Furthermore, while the exemplary embodiments illustrated herein show the various components of the system collocated in specific locations, it is to be appreciated that the various components of the system can be located or relocated at distant portions of a distributed network, such as a telecommunications network and/or the Internet, or within a dedicated secure, unsecured and/or encrypted system. Thus, it should be appreciated that the components of the system can be combined into one or more devices, such as a modem, or collocated on a particular node of a distributed network, such as a telecommunications network. As will be appreciated from the following description, and for reasons of computational efficiency, the components of the system can be arranged at any location within a distributed network without effecting the operation of the system. For example, the various components can be located in a central office (CO or ATU-C) modem, a customer premises modem (CPE or ATU-R), or some combination thereof. Similarly, the functionality of the system could be distributed between one or more of the modems and an associated computing device.

Furthermore, it should be appreciated that the various links, including communications link **20**, connecting the elements can be wired or wireless links, or any combination thereof or any other known or later-developed element(s) that is capable of supplying and/or communicating data to and from the connected elements. The term module as used herein can refer to any known or later-developed hardware, software, or combination of hardware and software that is capable of performing the functionality associated with that element. Furthermore, as used herein, the term "transmitter" has the same meaning as the term "transmitting modem" or "transmitting transceiver" and the term "receiver" has the same meaning as "receiving modem" or "receiving transceiver."

FIG. **1** illustrates an exemplary embodiment of an impulse noise adaptation system **100** according to an exemplary embodiment of this invention. In particular, the system **100** comprises a receiving modem **200** and a transmitting modem **300**. The exemplary receiving modem **200** comprises a decoder/deinterleaver **210**, a bit error rate (BER) detection module **220**, a forward error correction and interleaving parameter (FIP) module **230**, an impulse noise length determination module **240**, an impulse noise period determination module **250**, an impulse noise parameter management module **260**, an impulse noise protection adaptation module **270**, a synchronization module **280**, a message module **290** and a controller and memory (not shown) all interconnected by a link **5**. The exemplary transmitting modem **300** comprises a management module **310**, a synchronization module **320**, a message module **330**, and may optionally include an impulse noise parameter management module **340**.

In operation, the exemplary communication system adapts the impulse noise parameters on-line by operating using a series of different FIP settings. For each FIP setting, the system can dynamically determine if the appropriate amount of impulse noise protection is being provided. Based on these determinations, the system can select a particular FIP setting for regular, i.e., Showtime, operation. As will be discussed in greater detail hereinafter, this impulse noise protection adaptation can be performed during Showtime and/or during initialization.

Impulse noise protection adaptation during Showtime includes the following exemplary steps. First, the DSL system, i.e., the transmitting and receiving modems, completes regular initialization and commences transmission and reception using a first FIP setting. For DSL systems, this first FIP setting is selected by the receiver **200** and is based on the minimum/maximum data rate, maximum latency, and minimum INP parameters as configured by, for example, the service provider via the management module **310**. A detailed explanation of this procedure can be seen in G.992.3.

The system will use this first FIP setting for a period of time T**1**. During this period, and in conjunction with the BER detection module **220**, the receiver **200** detects if bit errors have occurred using this first FIP setting for the decoding and deinterleaving performed by the decoder/deinterleaver **210**. For example, the receiver **200** can use a CRC to detect bit errors. If there are no bit errors, then the current INP setting is adequate and there is no need to modify the INP setting and Showtime communication can continue as normal.

However, if bit errors have been detected by the BER detection module **220**, an adjustment to the INP setting may be appropriate. For example, a service provider, a user, or the system can choose an updated INP setting. Since bit errors have been detected, the INP setting could be increased by an on-line modification to the FIP parameters. Specifically, the transmitter **300**, and for example the management module **310**, in response to an indication from the receiver **200** that bit errors have been detected, can send a message to the receiver **200** to initiate a change in the INP settings. The receiver **200**, in cooperation with the message module **290**, the FIP module **230**, the impulse noise parameter management module **260** and impulse noise protection adaptation module **270**, can return a message to the transmitter **300** that specifies a newly determined FIP setting that satisfies the new INP requirement.

The transmitter **300** and receiver **200** can then transition to the new INP setting by starting to use the new FIP parameters

for transmission and reception, respectively, at a synchronized point in time. This synchronization can be done in accordance with a number of exemplary methods that are discussed hereinafter.

Upon synchronization, the system **100** continues communication using the updated FIP settings for a second period of time T**2**. If the receiver detects that bit errors are occurring using this updated FIP setting during the decoding and deinterleaving performed by the decoder/deinterleaver **210** the steps can be repeated with the selection of another updated INP setting. However, if there are no bit errors detected by the receiver **200** during the time period T**2**, then the new INP setting is adequate and there is no need for further change at which point the INP adaptation procedure can end.

This Showtime-based INP adaptation process can be repeated as many times as necessary until an INP setting is obtained that provides the required impulse noise immunity and/or bit error rate.

One advantage of this technique is that the transition between different FIP settings can be accomplished without reinitializing the transceivers using the lengthy standard initialization procedure that is typically used in ADSL and VDSL systems. In contrast, the transition can occur without the standard initialization since the transition between FIP settings can be synchronized between the transmitter and the receiver such that the receiver **200** can determine when to start FEC decoding using the new FIP settings for K and R. As will be discussed hereinafter, this transition can be synchronized using a number of different exemplary methods.

The transition can also be accomplished without synchronization in which case the receiver **200** will determine when the new FIP settings are used by some alternative means. For example, the receiver could determine when the new FIP settings are being used by FEC decoding using both the old and updated FIP settings and determining which one is currently being utilized by the transmitter by determining with which FIP setting the codeword is correct. Other receiver functions such as CRCs, ATM HEC errors and the like could also be used.

This impulse noise protection adaptation procedure can be performed during regular steady-state transmission, i.e., Showtime in ADSL, using actual user data or, for example, idle ATM cells. This methodology can also be performed during a special impulse noise training period during which the system is not actually transmitting user data. For example, this impulse noise training period could utilize transmitted predefined pseudorandom bit streams or, for example, idle ATM cells or HDLC flags for determining an appropriate INP value.

The length of the time periods T**1**, T**2**, . . . can be controlled by, for example, the receiving modem **200**. The receiving modem **200** could control the length of these time periods by, for example, sending a message that specifies how long the transmitter should transmit using a particular FIP setting. This length can be defined, for example, in terms of the number of DMT symbols or the number of FEC codewords. For example, a message could indicate that 200 DMT symbols should be sent for all FIP settings or 300 FEC code words should be sent for all FIP settings. Alternately, the message can indicate a different number of DMT symbols or FEC code words for each FIP setting. This technique could further be used to aid in determining an appropriate INP setting by forwarding a predetermined number of DMT symbols at a first FIP value, followed by a second number of DMT symbols at a second FIP value, and so on. For each of these sets of DMT symbols and associated FIP values, the receiver could determine the bit error rate, with the cooperation of the BER

detection module **220**, and then forward a message, with the cooperation of the message module **290**, to the transmitter **300** specifying which FIP setting provided the appropriate impulse noise protection.

Similarly, the transmitting modem **300** and receiving modem **200** could cooperate such that an optimum impulse noise protection value is converged upon. For example, the transmitting modem **300**, with the cooperation of the message module **330**, could forward a message to the receiving modem **200** specifying a first INP setting. If there are no bit errors, a message could be sent via the message module **290** to the transmitting modem **300** indicating that there are no bit errors and a lower INP value could be attempted. The impulse noise parameter management module **260** and FIP module **230** could then determine an appropriate lower INP setting and forward it to the transmitter **300**. As discussed above, since a lower INP value is inversely proportional to the user data rate, it is advantageous to optimize the INP value based on detected errors. The transmitting modem **300**, in cooperation with the message module **330**, could then send another message to the receiving modem **200** indicating that the lower INP value will be transitioned to at which point the procedure repeats itself through detection of bit errors in cooperation with the bit error rate detection module **220** and the decoder/deinterleaver **210**. This reduction in the INP parameters can continue until bit errors are detected at which point a message is sent, with the cooperation of the message module **290**, to the transmitting modem **300** indicating a higher INP value is required since bit errors were detected. Then, for example, based on an evaluation of a convergence of bit errors vs. impulse noise protection parameters, an optimum INP value can be determined and transitioned to by the receiver **300** and transmitter **200**. This procedure could also be used to increase the INP parameters in a similar fashion.

The length of the time periods T**1**, T**2**, . . . can also be controlled by the transmitter **300**. For example, the transmitter **300** could control the length of these time periods, by, for example, sending a message to the receiver **200** that specifies the length of time the transmitter will transmit using a particular FIP setting. This length could be defined in terms of the number of DMT symbols or, for example, the number of FEC codewords. The exemplary message could indicate that, for example, 200 DMT symbols would be sent for all FIP settings or 300 FEC codewords will be sent for all FIP settings. The message could also indicate a different number of DMT symbols or FEC codewords for each FIP setting. In general, the message could contain any type of information relating to how long the transmitter will be using a specific FIP setting(s).

The length of the time periods T**1**, T**2** . . . can also be controlled by, for example, a service provider or a user. For example, the service provider could configure through the management module **310** a minimum time X=10 seconds to be used for testing each FIP setting. The service provider could use the knowledge of the nature of the impulse noise, such as how often impulse noise event occurs, to determine the times. However, and in general, the length of the time periods could be configured to be any length of time as appropriate.

As discussed above, when a determination is made that there are bit errors and an increase in the INP setting is desirable, the receiving modem **200** can determine the new INP settings. However, it should be appreciated that the transmitting modem **300**, in cooperation with the impulse noise parameter management module **340** could also determine updated INP settings. For example, either modem could con-

tinually adapt the INP values as discussed above until no bit errors that are the result of impulse noise are detected by the BER detection module **220**.

As alluded to above, the receiver and transmitter can synchronize the modification of the FEC and interleaving parameters such that the both the transmitter and receiver start using the parameters at the same instant in time. This synchronization can be based on, for example, a synchronization using FEC codeword counters or a flag signal.

For synchronization using FEC codeword counters, the receiver **200** and transmitter **300** can synchronize the change in cooperation with the sync modules **280** and **320**, by counting the FEC codewords from the beginning of Showtime and the transition would occur when a specific FEC codeword counter value that is known by both the transmitter **300** and the receiver **200** is reached. Prior to the transition point, the transmitter **300** or receiver **200** in cooperation with the message module **290** and message module **330**, would send a message indicating the FEC codeword count value on which the FIP parameters will be updated. For example, the transmitting modem **300** can enter into Showtime and the synchronization module **320** commences counting the number of transmitted FEC codewords. For example, the first codeword transmitted has a count value of 0, the second transmitted codeword has a count value of 1 . . . . The counter can optionally be defined as having a finite length, for example, 0 to 1023 (10 bits) such that when the value of 1023 is reached, on the next FEC code word that is transmitted, the counter restarts at the value of 0.

Similarly, the receiving modem **200** upon entering Showtime starts a counter in cooperation with the synchronization module **280** that counts the number of received FEC codewords. The first received FEC codeword has a count value of 0, the second received FEC codeword to the count value of 1 . . . . As with the synchronization module **320**, the synchronization module **280** can have a counter with a finite length, for example 0-1023, such that when the value of 1023 is reached, on the next FEC codeword the counter restarts at 0.

At some point, for example, based on detected bit errors, the user, a service provider, or the like, it is determined that an updated FIP setting is needed. The receiving modem **200** can send a message, via the message module **290**, to the transmitting modem **300**, and in particular the message module **330** and impulse noise parameter management module **340**. The updated FIP setting can alternatively be sent from the transmitting modem to the receiving modem.

The synchronization module **280** and message module **290** then cooperate to send a message to the transmitting modem **300** specifying the FEC codeword counter value on which the new FIP settings are to be used for transmission and reception. Alternatively, the transmitting modem, in cooperation with the message module **330**, can send a message to the receiving modem indicating the FEC codeword counter value on which the FIP settings are to be used for transmission and reception. For example, the message can indicate that when the code word counter equals **501**, the new FIP setting will be used for transmission and reception.

When the transmitter FEC codeword counter equals the value indicated in the message, the synchronization module **220** instructs the transmitting modem **300** to transition to the new FIP settings. Similarly, when the synchronization module **280** in the receiving modem **200** counts the FEC codeword that equals the value indicated in the message, the receiving modem **200** transitions to using the new FIP settings for reception.

Synchronization can also be performed through the use of a flag signal (sync flag). For this exemplary embodiment, the receiving modem **200** and transmitting modem **300**, in cooperation with the synchronization module **280** and synchronization module **320**, synchronize the change in FIP settings using a flag or marker signal that is similar to that used in the ADSL2 G.992.3 ORL protocol. This protocol may be more desirable than using an FEC codeword counter because, for example, it has greater impulse noise immunity.

For synchronization using a flag signal, the receiver and transmitter would start using updated FEC and interleaving parameters on a pre-defined FEC codeword boundary following the sync flag. For example, while transmitting using a first INP setting, a determination is made by, for example, the BER detection module **220** that a new FIP setting is needed due to the presence of impulse noise on the line. This determination can be performed by the receiving modem, in cooperation with the FIP module **230**, the transmitting modem, a user, a service provider, or the like. The receiving modem **200**, in cooperation with the message module **290**, sends a message to the transmitting modem **300** indicating the new FIP settings to be used for transmission and reception. Alternatively, the transmitting modem, in cooperation with the message module **330**, prepares and sends a message to the receiving modem indicating the updated FIP settings to be used for transmission and reception.

The transmitting modem then sends a flag or marker signal to the receiving modem **200** indicating that the new FIP settings are to be used on a predetermined number of DMT symbols following the transmission of the flag or marker signal. For example, the flag signal could be an inverted sync symbol, or sync FLAG, as used in the ADSL2 G.992.3 OLR protocol. The transmitting modem **300** then starts using the new FIP settings for transmission on the predetermined number of DMT symbols following the transmission of the flag or marker signal. Similarly, the receiving modem starts using the new FIP settings for reception once the predetermined number of DMT symbols following the receipt of the flag or marker signal have been received.

EXAMPLE 1

On-Line INP Adaptation FIP Setting

This section describes an example of FIP settings for On-Line INP adaptation for DSL. In this example only the number of information bytes in a codeword (K) and the number of parity bytes in a codeword (R) are updated on-line. The Codeword Size (N) and Interleaver Depth (D) are not changed. This means that the latency (and interleaver memory size) and the line rate are not modified on-line. Since N=K+R this places restrictions on the allowed values for K and R.

1st Setting—{Approximate User Data Rate=3.968 Mbps, Line Rate=4.096 Mbps, N=128, K=124, R=4, S=1, D=64, Latency=16 msec, INP=1}

2nd Setting—{Approximate User Data Rate=3.840 Mbps, Line Rate=4.096 Mbps, N=128, K=120, R=8, S=1, D=64, Latency=16 msec, INP=2}

3nd Setting—{Approximate User Data Rate=3.584 Mbps, Line Rate=4.096 Mbps, N=128, K=112, R=16, S=1, D=64, Latency=16 msec, INP=4}

The On-line INP adaptation process is restricted to only modify the number of information bytes in a codeword (K) and the number of parity bytes in a codeword (R). The FEC Codeword Size (N=K+R) and Interleaver Depth (D) are not changed. This means that the latency (or interleaver memory size) and the line rate are not modified on-line. However the user data rate will change during the process since K is being modified. Also since the line rate and the FEC codeword size

are not modified, the S value does not change in the process. It is important to note that with these constraints, the On-line INP adaptation process can be done in as seamless manner (no bit errors and service interruption). This means provided that the modification if the FIP setting is restricted to K and R, the transition between FIP settings can be done in a seamless manner. This is the case because if the codeword size N and the interleaver depth D are not modified, the transition can happen without the problem of "interleaving memory flushing." Interleaver memory flushing is a well-known problem in which errors occur because interleaver and deinterleaver memory locations are overwritten due to on-line changes in the codeword size (N) and or interleaver depth (D).

EXAMPLE 2

On-Line INP Adaptation FIP Setting

This section describes an example of FIP settings for On-Line INP adaptation for DSL. In this example only the Codeword Size (N) and the number of parity bytes in a codeword (R) are updated on line. The number of information bytes in a codeword (K) and Interleaver Depth (D) are not changed and therefore the user data rate does not change. This means that the latency (and interleaver memory size) and the line rate are modified on-line. Since N=K+R this places restrictions on the allowed values for K and R.

1st Setting—{Approximate User Data Rate=3.968 Mbps, Line Rate=4.096 Mbps, N=128, K=124, R=4, S=1, D=64, Latency=16 msec, INP=1}

2nd Setting—{Approximate User Data Rate=3.968 Mbps, Line Rate=4.224 Mbps, N=132, K=124, R=8, S=1, D=64, Latency=16 msec, INP=2}

3nd Setting—{Approximate User Data Rate=3.968 Mbps, Line Rate=4.480 N=128, K=124, R=16, S=1, D=64, Latency=16 msec, INP=4}

In this example the Line Rate is modified on-line. For this reason it is necessary to also complete a rate adaptation process in order to complete this On-Line INP adaptation. A method for Seamless Rate Adaptation is described in U.S. Pat. No. 6,498,808.

While these examples restrict the changes to a subset of the FIP parameters, they can obviously be extended to cover any combination of the FIP parameters (N, K, R and D). For example, the value of D could also be modified in addition to the values of K, R and N. This could result in a change in the required interleaver memory and latency. In order to keep the memory and latency constant it is necessary to change the codeword size (N) accordingly when changing the interleaver depth (D). For example, if the interleaver depths is changed from D=64 to D=128, the Codeword size would have to be decreased by a factor of 2 so that overall latency is constant.

FIG. **2** illustrates an exemplary modified initialization state machine that includes the INPTraining state prior to Showtime. This state machine is based on the VDSL G.993.1 ITU standard. As defined in G.993.1, VTU-O is the VDSL Transceiver Unit at the Optical Network Unit (ONU) and VTU-R is the VDSL Transceiver Unit at the Remote terminal The Initialization state machine in FIG. **2** is an example of how INPTraining states could be included in a modified VDSL initialization procedure. While exemplary FIG. **1** includes the INPTraining state at a specific time in the initialization sequence, the INPTraining state can be included at any time during initialization provided that it is preceded by a state during which Showtime parameters are exchanged between the transceivers.

O-P-INPTrain

During the O-P-INPTrain State the VTU-O transmitter transmits DMT Symbols using the standard PMD, PMS-TC and TPS-TC SHOWTIME functions with parameters exchanged during the previous Exchange Phase. During this state, the TPS-TC transmits idle ATM cells, HDLC flags or 64/65 idle packets. SOC is inactive during this state.

This state is used by the VTU-R to determine the correct DS INP setting based on Impulse Noise conditions on the line. For example, the downstream CRC, FEC error detection, TPS-TC error detection capabilities, and other receiver functions can be used to determine whether the INP setting is adequate. The receiver can also use these receiver functions to determine the correct INP setting.

If the VTU-R determines that the current INP setting is not adequate, the VTU-R transmits the R-P-ISYNCHRO2 signal to indicate the need to transition back to the Exchange Phase in order to exchange new transmission parameters.

If the VTU-R determines that the current INP setting is not adequate, the VTU-R transmits the R-P-SYNCHRO2 signal to indicate that it is OK to transition to Showtime with the current transmission parameters.

R-P-INPTrain

During the R-P-INPTrain State the VTU-R transmitter transmits DMT Symbols using the standard PMD, PMS-TC and TPS-TC Showtime functions with parameters exchanged during the previous Exchange Phase. During this state, the TPS-TC transmits idle ATM cells, HDLS flags or 64/65 idle packets. SOC is inactive during this state.

This state is used by the VTU-O to determine the correct US INP setting based on Impulse Noise conditions on the line. For example, the upstream (US) CRC, FEC error detection, TPS-TC error detection capabilities, and other receiver functions can be used to determine whether the INP setting is adequate. The receiver can also use these receiver functions to determine the correct INP setting.

If the VTU-O determines that the current INP setting is not adequate, the VTU-O transmits the O-P-ISYNCHRO2 signal to indicate the need to transition back to the Exchange Phase in order to exchange new transmission parameters.

If the VTU-O determines that the current INP setting is not adequate, the VTU-O transmits the O-P-SYNCHRO2 signal to indicate that it is OK to transition to Showtime with the current transmission parameters.

O-P-SYNCHRO2

The O-P-SYNCHRO2 is the same as defined in the current VDSL1. As in VDSL1, the VTU-O transmitter enters Showtime after transmitting the O-P-SYNCHRO2 signal.

But, if the VTU-R has not also entered into Showtime, the VTU-O waits for receipt of the R-P-SYNCHRO2 or R-P-ISYNCHRO2. If the VTU-O receives the R-P-SYNCHRO2 it continues in Showtime. If the VTU-O receives the R-P-ISYNCHRO2, the VTU-O transmitter transitions back to the beginning of the O-P-MEDLEY state.

R-P-SYNCHRO2

The O-P-SYNCHRO2 is the same as defined in the current VDSL1. As in VDSL1, the VTU-R transmitter enters Showtime after transmitting the R-P-SYNCHRO2 signal.

But, if the VTU-O has not also entered into Showtime, the VTU-R waits for receipt of the O-P-SYNCHRO2 or O-P-ISYNCHRO2. If the VTU-R receives the O-P-SYNCHRO2 it continues in Showtime. If the VTU-R receives the O-P-ISYNCHRO2, the VTU-R transmitter transitions back to the beginning of the R-P-MEDLEY state.

O-P-ISYNCHRO2

The O-P-ISYNCHRO2 is phase-inverted version of the O-P-SYNCHRO2, i.e., a subcarrier-by-subcarrier 180

degrees phase reversal. The VTU-O transmitter transitions to the beginning of the O-P-MEDLEY state after transmitting the O-P-ISYNCHRO2 signal.

R-P-ISYNCHRO2

The R-P-ISYNCHRO2 is phase-inverted version of the R-P-SYNCHRO2, i.e., a subcarrier-by-subcarrier 180 degrees phase reversal. The VTU-R transmitter transitions to the beginning of the R-P-MEDLEY state after transmitting the R-P-ISYNCHRO2 signal

Exemplary Overview State Transition Rules Based on SYNCHRO2 and the ISYNCHRO2 signals:

If either the VTU-O or the VTU-R transmits the ISYNCHRO2 signal then both VTU-R and VTU-O transition back to beginning of the MEDLEY state.

If both the VTU-O and the VTU-R transmit the SYNCHRO2 signal then both the VTU-R and the VTU-O transition into Showtime.

There are several important points regarding this exemplary embodiment. First, the receiver can measure the efficiency of the plurality of INP settings without completing a new initialization procedure such as is used in ADSL and VDSL systems. Second, the length of the time of the O-P-INPTrain and R-P-INPTrain states can be controlled by the VTU-R and VTU-O respectively. In this way the receivers have adequate time to determine if the current INP setting is correct. In order to accomplish this, prior to entering the R-P-INPTrain state, the VTU-O can transmit a message to the VTU-R indicating the minimum length of the R-P-INPTrain state. Likewise, prior to entering the O-P-INPTrain state, the VTU-R can transmit a message to the VTU-O indicating the minimum length of the O-P-INPTrain state. For example, the message could indicate that a minimum of 20000 DMT symbols should be sent during the O-P-INPTrain state. Additionally, the length of the time of the O-P-INPTrain and R-P-INPTrain states could be set by the DSL service provider in order to make sure that the initialization does not take too long or because the service provider may have some knowledge of the statistics of the impulse noise which require setting the length of the INPTrain states. In this exemplary case a message could be sent from the VTU-O to the VTU-R indicating the minimum and/or maximum length of the O-P-INPTrain and/or the R-P-INPTrain states. For example, the message could indicate that a minimum of 20000 DMT symbols should be sent during the R-P-INPTrain state. Also, for example, the message could indicate that a maximum of 40000 DMT symbols should be sent during the R-P-INPTrain state.

The length of the time of the O-P-INPTrain and R-P-INPTrain states could also be controlled by the VTU-O and VTU-R respectively. In this case, the transmitters will have control of the state lengths. In order to accomplish this, prior to entering the R-P-INPTrain state, the VTU-R can transmit a message to the VTU-O indicating the minimum length of the R-P-INPTrain state. Likewise, prior to entering the O-P-INPTrain state, the VTU-O can transmit a message to the VTU-R indicating the minimum length of the O-P-INPTrain state. For example, the message could indicate that a minimum of 20000 DMT symbols should be sent during the O-P-INPTrain state.

As discussed above, another exemplary aspect of this invention relates to determining the length and/or repetition period of impulse noise events in order to select an appropriate INP setting.

The INP length can be determined using any one of a plurality of metrics. For example, the INP length can be determined based on one or more of:

1) The length of the impulse in time
2) The number of received bits that are affected by the impulse noise
3) The number of received ATM cells that are affected by the impulse noise
4) The number of 64/65 packets that are affected by the impulse noise
5) The number of received DMT symbols that are affected by the impulse noise
6) The number of FEC codewords that are affected by the impulse noise

As an example, assume that length of the impulse noise is 700 microseconds (Example 1). At a data rate of 1 Mbps this could correspond to an impulse noise length of 700 bits (Example 2) assuming that all impulse noise was high enough to affect all 700 bits in the 700 microsecond period. This also could correspond to a impulse noise length of 700/(53*8) =1.65 ATM cells since there are 53 bytes in a ATM cell (Example 3). This also could correspond to an impulse noise length of 700/(65*8)=1.35 64/65 packets since there are 65 bytes in a 64/65 packet (Example 4). This also could correspond to an impulse noise length of 700/(250)=2.8 DMT symbols (INP=2.8) assuming the DMT symbol rate is 4 kHz (250 microseconds DMT symbol length) (Example 5).

As an example, the determination of the INP length with the number of received bits that are affected by impulse noise can be performed in accordance with the following procedure. Initially, the transmitting modem **300** transmits data using at least one of the Showtime functions. In accordance with a first exemplary embodiment, at least the bit allocation table is used. In addition, or alternatively, the trellis coder, framer, and TPS-TC functions may also be used. Additionally, or alternatively still, the interleaving and FEC may also be used.

The receiving modem **200** receives data using at least one of the Showtime functions, such as the bit allocation table as discussed above. Similarly, the trellis coder, framer, TPS-TC and/or interleaving and FEC can be used.

The transmitter **300** then transmits a predefined bit pattern that is used for determining or measuring the impulse noise length based on detected erroneous bits. The receiving modem **200** compares, with the assistance of the impulse noise length determination module **240**, the predefined bit pattern to the received bit pattern in order to detect bit errors. Since impulse noise events typically cause a burst of errors in a bit stream, the receiving modem **200** determines the length of the impulse noise event by detecting and determining the length of the error burst.

Once the receiver **200** determines the length of the impulse noise, the receiving modem **200**, in cooperation with the message module **290**, sends a message to the transmitter **300** that indicates the determined length of the impulse noise.

In accordance with an exemplary embodiment, the FEC correction capability and interleaving is turned off when trying to determine the length of the impulse noise, e.g., R=0 and D=1. Disabling the FEC and interleaving is beneficial when trying to determine the length of the impulse noise based on affected bits, affected code words, or affected packets. This is the case because if the FEC/interleaving is enabled, the impulse event can be spread over a large time period which makes it more difficult to determine the length of the original impulse.

The exemplary methodology discussed above can also be used to determine the length of the impulse event using other metrics. For example, a predefined bit pattern could be used in the payload of the ATM cells or the 64/65 packets. Idle packets or cells, which carry a predefined pattern, could also be used. In these cases, the receiver would compare the received

data to the predefined data to determine the length of the impulse or to determine if an INP setting is adequate. Alternatively, the CRC of the ATM cell or 64/65 packet could be used to determine the length of the impulse noise. This would provide a courser measurement of the impulse noise length since it would be an integer number of packets or cells. In this example, a 700 bit impulse would cause CRC errors in two ATM cells so the length of the impulse noise would be two ATM cells, as opposed to the more precise measurement of 1.64 above. Likewise, in this example, a 700 bit impulse would cause CRC errors in two 64/65 packets so the length of the impulse noise would be two packets, as opposed to the more precise measure of 1.35 above.

Likewise, a predefined bit pattern could also be used to modulate the carriers in the DMT symbols so that the receiver would know what DMT symbols were transmitted. In this case, the receiver **200** would determine how many DMT symbols were corrupted by comparing the received signal with a known transmitted signal.

The exemplary techniques used to determine the length of the impulse noise event could also be extended to determine the repetition rate of the impulse noise event in cooperation with the impulse noise period determination module **250**. In order to determine the repetition period of an impulse noise event, the receiver **200** detects the impulse noise event as discussed above and then determines how often they occur. For example, periodic impulse noise due to AC power lines occur at 120 Hz reception rates or approximately every 8 ms. The receiver **200** could, for example, also store information about past impulse noise events and compare detected impulse noise events to historical events. The impulse noise period determination module **250** could then determine which events of a similar duration are occurring at what interval. This information could then be used in determining an appropriate INP setting.

Once the receiver **200** determines the (maximum) length of the impulse and/or the repetition period of the impulse, the information, with the cooperation of the message module **290** can be sent to the transmitting modem. For example, when the receiving modem **200** determines the (maximum) length of the impulse and/or the repetition period of the impulse, the impulse noise period determination module **250**, in cooperation with the message module **290** can forward information to the transmitting modem that quantifies this period. The transmitting modem **300** could provide this information to, for example, the management module **310**, that would allow, for example, an operator or service provider to configure the modems. For example, based on the period information contained in the message, the operator may configure the modems to different INP values, data rates, latency, or the like.

The receiver **200** could also test a specific INP setting by detecting how many received bits are errored in a specific time period. For example, if the specific INP setting enables the correction of 100 bits in an 8 msec. time period then the receiver **200** could detect how many bits are errored in a sliding 8 msec. window. If less than 100 bits are detected in error in the 8 ms sliding window, then the INP setting is adequate.

If more than 100 bits are detected in error in a sliding window, then the INP setting is not adequate, and the FEC/interleaving needs to be changed to provide more error correction capability.

Likewise, instead of using the received number of bits that are effected by the impulse noise, the receiver could test a specific INP setting by detecting how many received ATM cells, 64/65 packets, DMT symbols, and/or FEC corrections are affected in a specific time period.

FIG. **3** outlines an exemplary method for performing impulse noise protection adaptation during Showtime according to this invention. In particular, control begins in step **S300** and continues to step **S310**. In step **S310**, traditional DSL initialization occurs. Next, in step **S320**, Showtime is entered between the two modems using the first FIP setting that was determined during the initialization in step **S310**. Then, in step **S330**, a determination is made whether bit errors are occurring using the first FIP setting. If bit errors are not occurring, control continues to step **S340** where the control sequence ends. Otherwise, control jumps to step **S350**.

In step **S350**, a determination is made that an increase of the INP setting is required that requires modification of the FIP parameters. Next, in step **S360**, updated INP parameter is determined and a message forwarded to the receiver specifying the new INP setting. Then, in step **S370**, the receiver forwards to the transmitter updated FIP parameters for the new impulse noise protection requirements. Control then continues to step **S380**.

In step **S380**, the transmitter and receiver transition to using the updated INP parameters at a synchronization point. Next, in step **S390** Showtime operation continues. Control then continues back to step **S330**.

FIG. **4** outlines an exemplary method for receiver optimized impulse noise protection adaptation during Showtime. In particular, control begins in step **S400** and continues to step **S410**. In step **S410**, the DSL system completes regular initialization In particular, control begins in step **S400** and continues to step **S410**. In step **S410**, the DSL system completes startup initialization and continues into Showtime in step **S420** using a first FIP setting. Control then continues to step **S430** where a determination is made whether bit errors are occurring using the first FIP setting. If bit errors are not occurring, control continues to step **S440** where the control sequence ends.

Otherwise, control jumps to step **S450** where the receiver increases the impulse noise protection by modifying the FIP parameters. Next, in step **S460**, a synchronization point is determined between the transmitter and receiver, and when the synchronization point is reached both the transmitter and the receiver transition to the updated FIP setting in step **S470** and Showtime communications continue and control returns back to step **S430**.

FIG. **5** illustrates an exemplary method for synchronization of the modified FEC and interleaving parameters according to this invention. In particular, control begins in step **S500** and continues to step **S510**. In step **S510**, the transmitter enters Showtime and counts the number of transmitted FEC codewords. Next, in step **S520**, the receiver enters Showtime and counts the number of received FEC codewords. Then, in step **S530**, after a determination is made that an updated FIP setting is needed, the receiving modem sends a message to the transmitting modem indicating a new FIP setting that is to be used for transmitting and reception or, alternatively, the transmitting modem sends a message to the receiving modem indicating the new FIP setting to be used for transmission and reception. Control then continues to step **S540**.

In step **S540**, a message with the FEC codeword counter value on which the new FIP values are to be used is exchanged. Next, in step **S550**, a determination is made whether the counter value has been reached at the transmitter. If the counter value has not been reached, the next codeword is counted in step **S560** and control continues back to step **S550**.

Otherwise, if the counter value has been reached, control jumps to step S570. In step S570, the transmitter transitions to the new FIP setting. Next, in step S580, a determination is made whether the counter value has been reached at the receiving modem. If the counter value has not been reached, the next received FEC codeword is counted in step S585 and control continues back to step S580. However, when the counter value has been reached in the receiving modem, control jumps to step S590 where the receiving modem switches to the new FIP value. Control then continues to step S595 where the modems continue Showtime communication and control continues to step S597 where the control sequence ends.

FIG. 6 illustrates an exemplary method of synchronization using a flag signal according to this invention. In particular, control begins in step S600 and continues to step S610. In step S610, the modems enter Showtime using the first FIP parameters. Next, in step S620, a message is exchanged indicating the new FIP settings. Then, in step S630, the transmitter forwards to the receiver a flag signal indicating when the new FIP settings are to be used.

At step S640, and at a predefined change time following the transmission of the flag signal, the transmitter begins transmission using the new FIP parameters. Next, at step S650, at the predefined change time following the reception of the flag signal, the receiver commences reception utilizing the new FIP parameters. Control then continues to step S660 where Showtime communication continues with the control sequence ending at step S670.

FIG. 7 illustrates an exemplary method of impulse noise length and period determination. In particular, control begins in step S700 and continues to step S710. In step S710, the transmitter transmits data using at least one Showtime function. Next, in step S720, the receiver receives data using at least one Showtime function. Then, in step S730, the transmitter transmits predefined information to the receiver. Control then continues to step S740.

In step S740, the receiver receives the predefined information from the transmitter. Next, in step S750, the receiver compares the received predefined information to the predefined information and determines the differences (i.e., errors) between the two. Then, in step S760, and based on the detected errors, the length of the burst error is determined Next, in step S770, a message is forwarded to the transmitter indicating the length of the impulse noise event. Control then continues to step S780 where a determination is made whether the period of the impulse noise event is also to be determined If the period is not to be determined, control continues to step S790 where the control sequence ends. Otherwise, control jumps to step S800.

In step S800, and once the length of the impulse noise event is know, the receiver detects how often the impulse noise events occur. For example, historical information regarding the length and timing of previous impulse noise events can be stored in a memory (not shown). Then, a comparison can be made with the aid of a processor (not shown) to compare an impulse noise event to the historical information to determine the period of repetition (if any) of similar impulse noise events and, for example, a message indicating the period as well as the timing forwarded to, for example, another transceiver, the CO, or in general any destination as appropriate. Thus, in a similar manner, this information can be forwarded in step S810 to, for example, the transmitter in a message specifying the repetition frequency of the impulse noise event. Control then continues to step S820 where the control sequence ends.

The above-described system can be implemented on wired and/or wireless telecommunications device(s), such a modem, a multicarrier modem, a DSL modem, an ADSL modem, an xDSL modem, a VDSL modem, a linecard, test equipment, a multicarrier transceiver, a wired and/or wireless wide/local area network system, a satellite communication system, a modem equipped with diagnostic capabilities, or the like, or on a separate programmed general purpose computer having a communications device.

Additionally, the systems, methods and protocols of this invention can be implemented on a special purpose computer, a programmed microprocessor or microcontroller and peripheral integrated circuit element(s), an ASIC or other integrated circuit, a digital signal processor, a hard-wired electronic or logic circuit such as discrete element circuit, a programmable logic device such as PLD, PLA, FPGA, PAL, modem, transmitter/receiver, or the like. In general, any device capable of implementing a state machine that is in turn capable of implementing the methodology illustrated herein can be used to implement the various communication methods, protocols and techniques according to this invention.

Furthermore, the disclosed methods may be readily implemented in software using object or object-oriented software development environments that provide portable source code that can be used on a variety of computer or workstation platforms. Alternatively, the disclosed system may be implemented partially or fully in hardware using standard logic circuits or VLSI design. Whether software or hardware is used to implement the systems in accordance with this invention is dependent on the speed and/or efficiency requirements of the system, the particular function, and the particular software or hardware systems or microprocessor or microcomputer systems being utilized. The communication systems, methods and protocols illustrated herein however can be readily implemented in hardware and/or software, or any means, using any known or later developed systems or structures, devices and/or software by those of ordinary skill in the applicable art from the functional description provided herein and with a general basic knowledge of the computer and telecommunications arts.

Moreover, the disclosed methods may be readily implemented in software, that can be stored on a storage medium, and executed on programmed general-purpose computer, a special purpose computer, a microprocessor, or the like. In these instances, the systems and methods of this invention can be implemented as a program embedded on personal computer such as JAVA® or CGI script, as a resource residing on a server or computer workstation, as a routine embedded in a dedicated communication system or system component, or the like. The system can also be implemented by physically incorporating the system and/or method into a software and/or hardware system, such as the hardware and software systems of a communications transceiver.

It is therefore apparent that there has been provided, in accordance with the present invention, systems and methods for impulse noise adaptation. While this invention has been described in conjunction with a number of embodiments, it is evident that many alternatives, modifications and variations would be or are apparent to those of ordinary skill in the applicable arts. Accordingly, it is intended to embrace all such alternatives, modifications, equivalents and variations that are within the spirit and scope of this invention.

The invention claimed is:

1. In a transceiver, a method of adapting forward error correction and interleaver parameter (FIP) settings of the transceiver during steady-state communication or initialization comprising:

transmitting, by the transceiver, a signal using a first FIP setting;

transmitting, by the transceiver, a flag signal; and

switching, by the transceiver, to using, for transmitting, a second FIP setting following transmission of the flag signal,

wherein:

the first FIP setting comprises at least one first FIP value,

the second FIP setting comprises at least one second FIP value, different than the first FIP value, and

the switching occurs on a pre-defined forward error correction codeword boundary following the flag signal.

2. The method of claim **1**, wherein a first forward error correction parameter value of the first FIP setting is different than a second forward error correction parameter value of the second FIP setting.

3. The method of claim **1**, wherein a first interleaver parameter value of the first FIP setting is different than a second interleaver parameter value of the second FIP setting.

4. The method of claim **1**, wherein the adapting of the FIP settings is based on a service provider configuration.

5. The method of claim **1**, further comprising transmitting the flag signal on a telephone line that experiences impulse noise from external sources including one or more of AM radio, HAM radio and AC power lines.

6. The method of claim **1**, wherein the method is performed in a linecard that includes a management interface that is used by an operator or service provider to configure a service.

7. The method of claim **1**, wherein the method is performed in a Customer Premises Equipment (CPE) that includes a management interface that is used by an operator, a service provider or service user.

8. An apparatus configurable to adapt forward error correction and interleaver parameter (FIP) settings during steady-state communication or initialization comprising:

a transceiver, including a processor, configurable to:

transmit a signal using a first FIP setting,

transmit a flag signal, and

switch to using for transmission, a second FIP setting following transmission of the flag signal,

wherein:

the first FIP setting comprises at least one first FIP value,

the second FIP setting comprises at least one second FIP value, different than the first FIP value, and

the switching occurs on a pre-defined forward error correction codeword boundary following the flag signal.

9. The apparatus of claim **8**, wherein a first forward error correction parameter value of the first FIP setting is different than a second forward error correction parameter value of the second FIP setting.

10. The apparatus of claim **8**, wherein a first interleaver parameter value of the first FIP setting is different than a second interleaver parameter value of the second FIP setting.

11. The apparatus of claim **8**, wherein the adapting of the FIP settings is based on a service provider configuration.

12. The apparatus of claim **8**, wherein the transceiver is operable to transmit the flag signal on a telephone line that experiences impulse noise from external sources including one or more of AM radio, HAM radio and AC power lines.

13. The apparatus of claim **8**, wherein the transceiver is located in a linecard that includes a management interface that is usable by an operator or service provider to configure a service.

14. The apparatus of claim **8**, wherein the transceiver is located in a Customer Premises Equipment (CPE) that includes a management interface that is usable by an operator, a service provider or service user.

15. The apparatus of claim **8**, wherein the transceiver includes at least one digital signal processor.

16. The apparatus of claim **8**, wherein the transceiver includes at least one ASIC.

17. In a transceiver, a method of adapting forward error correction and interleaver parameter (FIP) settings of the transceiver during steady-state communication or initialization comprising:

receiving, by the transceiver, a signal using a first FIP setting;

receiving, by the transceiver, a flag signal; and

switching, by the transceiver, to using, for receiving, a second FIP setting following reception of the received flag signal,

wherein:

the first FIP setting comprises at least one FIP value,

the second FIP setting comprises at least one second FIP value, different than the first FIP value, and

the switching occurs on a pre-defined forward error correction codeword boundary following the flag signal.

18. The method of claim **17**, wherein a first forward error correction parameter value of the first FIP setting is different than a second forward error correction parameter value of the second FIP setting.

19. The method of claim **17**, wherein a first interleaver parameter value of the first FIP setting is different than a second interleaver parameter value of the second FIP setting.

20. The method of claim **17**, wherein the adapting of the FIP settings is based on a service provider configuration.

21. The method of claim **17**, further comprising receiving the flag signal over a telephone line that experiences impulse noise from external sources including one or more of AM radio, HAM radio and AC power lines.

22. The method of claim **17**, wherein the method is performed in a linecard that includes a management interface that is used by an operator or service provider to configure a service.

23. The method of claim **17**, wherein the method is performed in a Customer Premises Equipment (CPE) that includes a management interface that is used by an operator, a service provider or service user.

24. An apparatus configurable to adapt forward error correction and interleaver parameter (FIP) settings during steady-state communication or initialization comprising:

a transceiver, including a processor, configurable to:

receive a signal using a first FIP setting,

receive a flag signal, and

switch to using for reception, a second FIP setting following reception of the flag signal,

wherein:

the first FIP setting comprises at least one first FIP value,

the second FIP setting comprises at least one second FIP value, different than the first FIP value, and

the switching occurs on a pre-defined forward error correction codeword boundary following the flag signal.

25. The apparatus of claim **24**, wherein a first forward error correction parameter value of the first FIP setting is different than a second forward error correction parameter value of the second FIP setting.

26. The apparatus of claim **24**, wherein a first interleaver parameter value of the first FIP setting is different than a second interleaver parameter value of the second FIP setting.

27. The apparatus of claim **24**, wherein the adapting of the FIP settings is based on a service provider configuration.

28. The apparatus of claim **24**, wherein the transceiver is operable to receive the flag signal over a telephone line that

experiences impulse noise from external sources including one or more of AM radio, HAM radio and AC power lines.

29. The apparatus of claim **24**, wherein the transceiver is located in a linecard that includes a management interface that is usable by an operator or service provider to configure a service.

30. The apparatus of claim **24**, wherein the transceiver is located in a Customer Premises Equipment (CPE) that includes a management interface that is usable by an operator, a service provider or service user.

31. The apparatus of claim **24**, wherein the transceiver includes at least one digital signal processor.

32. The apparatus of claim **24**, wherein the transceiver includes at least one ASIC.

* * * * *

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(12) **United States Patent**
Tzannes

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(54) **IMPULSE NOISE MANAGEMENT**

(71) Applicant: **TQ Delta, LLC**, Austin, TX (US)

(72) Inventor: **Marcos C. Tzannes**, Orinda, CA (US)

(73) Assignee: **TQ Delta, LLC**, Austin, TX (US)

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(63) Continuation of application No. 12/769,193, filed on Apr. 28, 2010, now Pat. No. 8,462,835, which is a continuation of application No. 10/597,482, filed as application No. PCT/US2005/006842 on Mar. 3, 2005, now abandoned.

(60) Provisional application No. 60/549,804, filed on Mar. 3, 2004, provisional application No. 60/555,982, filed on Mar. 24, 2004.

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H04L 5/16 (2006.01)

(52) **U.S. Cl.**
USPC **375/219**; 375/224; 375/284; 375/346

(58) **Field of Classification Search**
USPC 375/219, 224, 285, 350, 774, 786
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

3,699,457 A	10/1972	Wright	
4,845,466 A	7/1989	Hariton et al.	
5,537,437 A	7/1996	Kaku et al.	
5,794,136 A	8/1998	Buchwald et al.	
6,498,808 B1	12/2002	Tzannes	
6,553,534 B2 *	4/2003	Yonge et al.	714/774
6,647,070 B1	11/2003	Shalvi et al.	
6,654,410 B2	11/2003	Tzannes	
6,683,855 B1	1/2004	Bordogna et al.	
6,788,923 B2	9/2004	Tsujishita et al.	
6,920,194 B2	7/2005	Stopler et al.	
7,050,419 B2	5/2006	Azenkot et al.	

(Continued)

OTHER PUBLICATIONS

ITU-T, G.992.3, Series G: Transmission Systems and Media, Digital Systems and Networks, International Telecommunication Union, Jul. 2002.

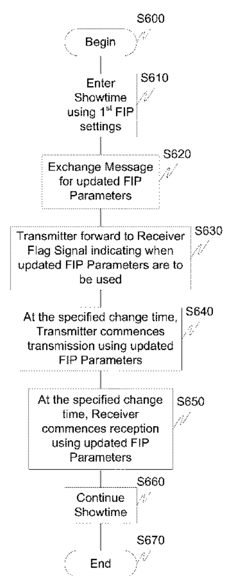
(Continued)

Primary Examiner — Jean B Corrielus
(74) *Attorney, Agent, or Firm* — Jason H. Vick; Sheridan Ross, PC

(57) **ABSTRACT**

Evaluation of the impact of impulse noise on a communication system can be utilized to determine how the system should be configured to adapt to impulse noise events. Moreover, the system allows for information regarding impulse noise events, such as length of the event, repetition period of the event and timing of the event, to be collected and forwarded to a destination. The adaptation can be performed during one or more of Showtime and initialization, and can be initiated and determined at either one or more of a transmitter and a receiver.

17 Claims, 7 Drawing Sheets



(56) **References Cited**

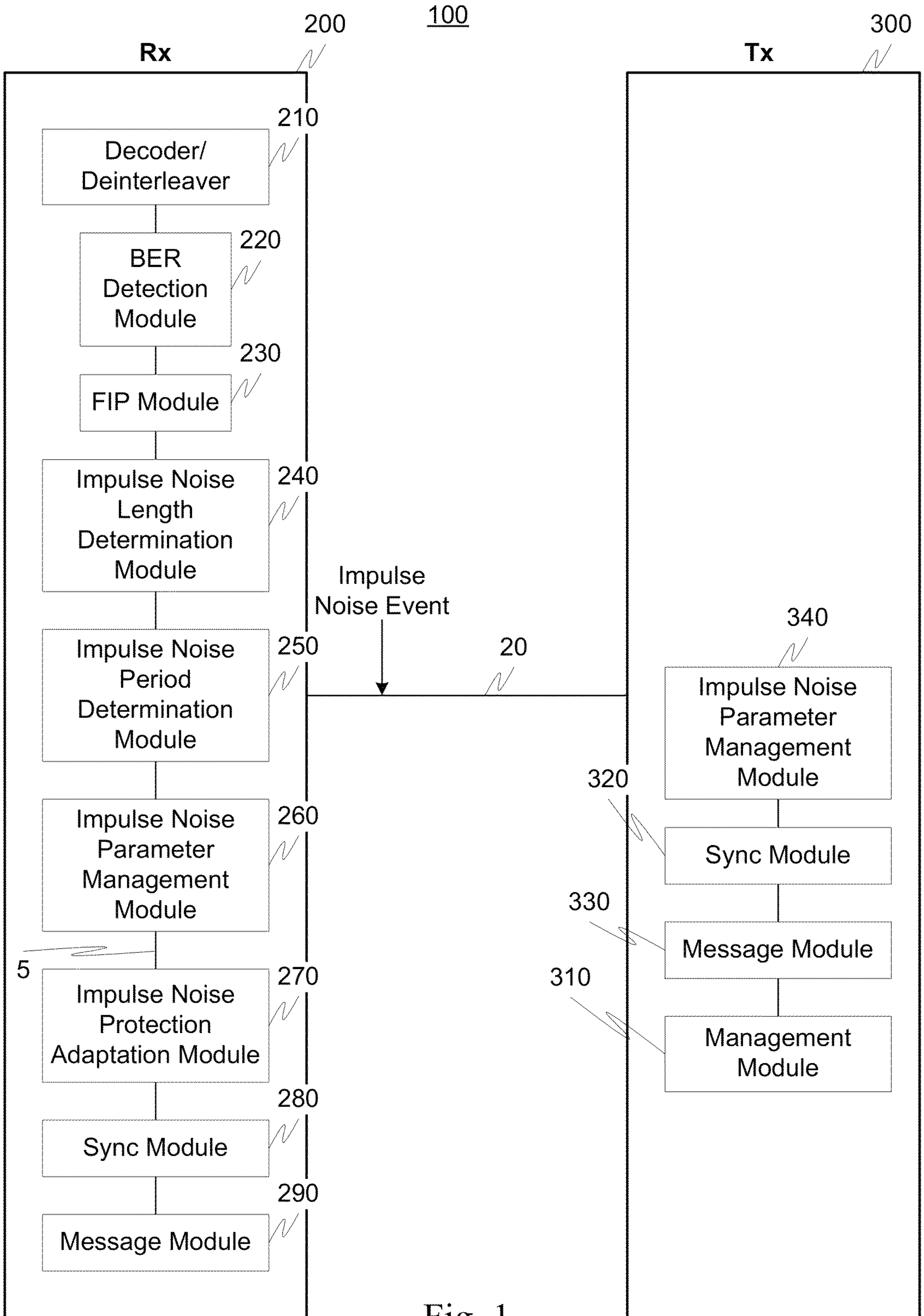
U.S. PATENT DOCUMENTS

7,103,820	B2 *	9/2006	Polk et al.	714/752
7,103,828	B2 *	9/2006	Polk et al.	714/776
7,162,676	B2	1/2007	Coleman et al.	
7,428,669	B2	9/2008	Cioffi	
7,673,221	B2	3/2010	Li et al.	
8,462,835	B2	6/2013	Tzannes	
2002/0069038	A1	6/2002	Cooper	
2002/0150167	A1	10/2002	Demjanenko et al.	
2005/0138524	A1 *	6/2005	Cioffi	714/758
2006/0078044	A1	4/2006	Norrell et al.	
2008/0232444	A1	9/2008	Tzannes	
2009/0086800	A1	4/2009	Cioffi	
2010/0229075	A1	9/2010	Yousef et al.	

OTHER PUBLICATIONS

ITU-T, G.993.1, Series G: Transmission Systems and Media, Digital Systems and Networks, International Telecommunication Union, Revision 2004.
Proposed revisions to G.992.4 and G.992.3&5 regarding impulse noise protection, ITU—Telecommunication Standardization Sector, SBC, BellSouth, BT, Singapore, Jan. 19-Jan. 24, 2004, 4 pages.
Moulin et al., "Discrete-multitone-based ADSL and VDSL systems performance analysis in an impulse noise environment—Advances in Communication Cables Technology," IEE Proceedings, Nov. 3, 2003, vol. 150(6), pp. 273-278.
Toumpakaris et al., "A Simple Byte-Erasure Method for Improved Impulse Immunity in DSL," 2003 IEEE International Conference on Communications, May 2003, vol. 1 of 5, pp. 2426-2430.
Invitation to Pay Additional Fees for International (PCT) Patent Application No. PCT/US2005/006842, mailed Jul. 28, 2005.
International Search Report for International (PCT) Patent Application No. PCT/US2005/006842, mailed Oct. 5, 2005.
Written Opinion for International (PCT) Patent Application No. PCT/US2005/006842, mailed Oct. 5, 2005.
International Preliminary Report on Patentability for International (PCT) Patent Application No. PCT/US2005/006842, dated Sep. 14, 2006.
Examiner's Report for Canadian Patent Application No. 2,555,757, mailed Oct. 17, 2012.
First Office Action for Chinese Patent Application No. 200580006738.8, mailed Oct. 23, 2009.
Second Office Action (including translation) for Chinese Patent Application No. 200580006738.8, issued Jun. 30, 2011.
Official Communication for European Patent Application No. 05724394.1, dated Aug. 19, 2010.
European Search Report for European Patent Application No. 10006702.4, dated Aug. 6, 2010.
Official Action for Indian Patent Application No. 2155/KOLNP/2006, dated Feb. 18, 2011.
Restriction Requirement for U.S. Appl. No. 10/597,482, mailed May 7, 2009.
Official Action for U.S. Appl. No. 10/597,482, mailed Jun. 30, 2009.
Official Action for U.S. Appl. No. 10/597,482, mailed Dec. 2, 2009.
Official Action for U.S. Appl. No. 10/597,482, mailed Jul. 19, 2010.
Official Action for U.S. Appl. No. 10/597,482, mailed Feb. 9, 2011.
Official Action for U.S. Appl. No. 12/769,193, mailed Sep. 22, 2010.
Official Action for U.S. Appl. No. 12/769,193, mailed Apr. 19, 2011.
Official Action for U.S. Appl. No. 12/769,193, mailed May 25, 2011.
Official Action for U.S. Appl. No. 12/769,193, mailed Jan. 22, 2013.
Notice of Allowance for U.S. Appl. No. 12/769,193, mailed Apr. 29, 2013.

* cited by examiner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200
Rx
210
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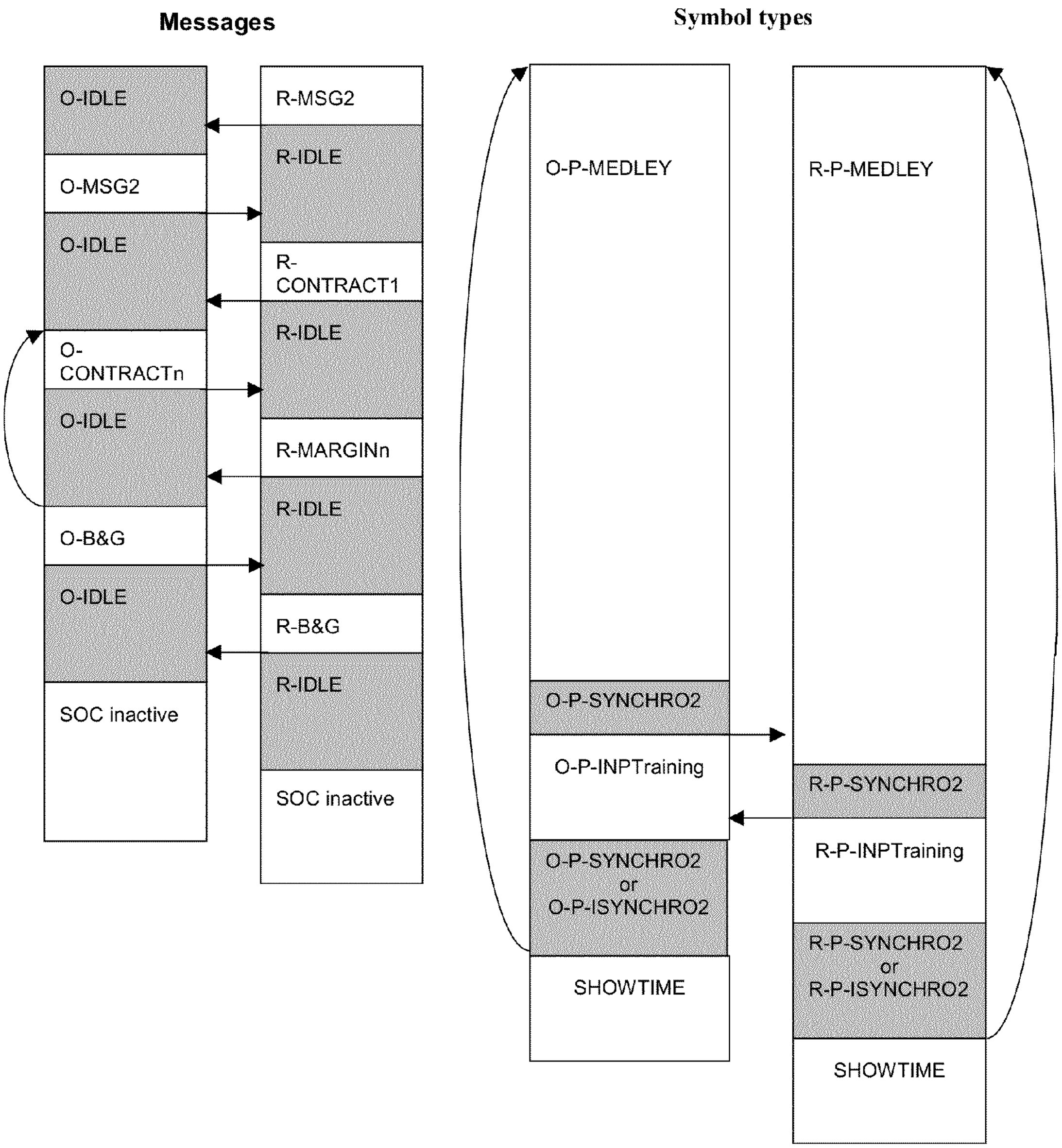
BER Detection Module
230
FIP Module
240
Impulse Noise Length Determination Module
250
Impulse Noise Period Determination Module
260
Impulse Noise Parameter Management Module
5
270
Impulse Noise Protection Adaptation Module
280
Sync Module
290
Message Module
Impulse Noise Event
20
300
Tx
340
Impulse Noise Parameter Management Module
320
Sync Module
330
Message Module
310
Management Module

Fig. 1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R-MARGINn
R-IDLE
O-B&G
O-IDLE
R-B&G
R-IDLE
SOC inactive
SOC inactive
O-P-MEDLEY
R-P-MEDLEY
O-P-SYNCHRO2
O-P-INPTraining
R-P-SYNCHRO2
R-P-INPTraining
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SHOWTIME
R-P-SYNCHRO2 or R-P-ISYNCHRO2
SHOWTIME

Fig. 2

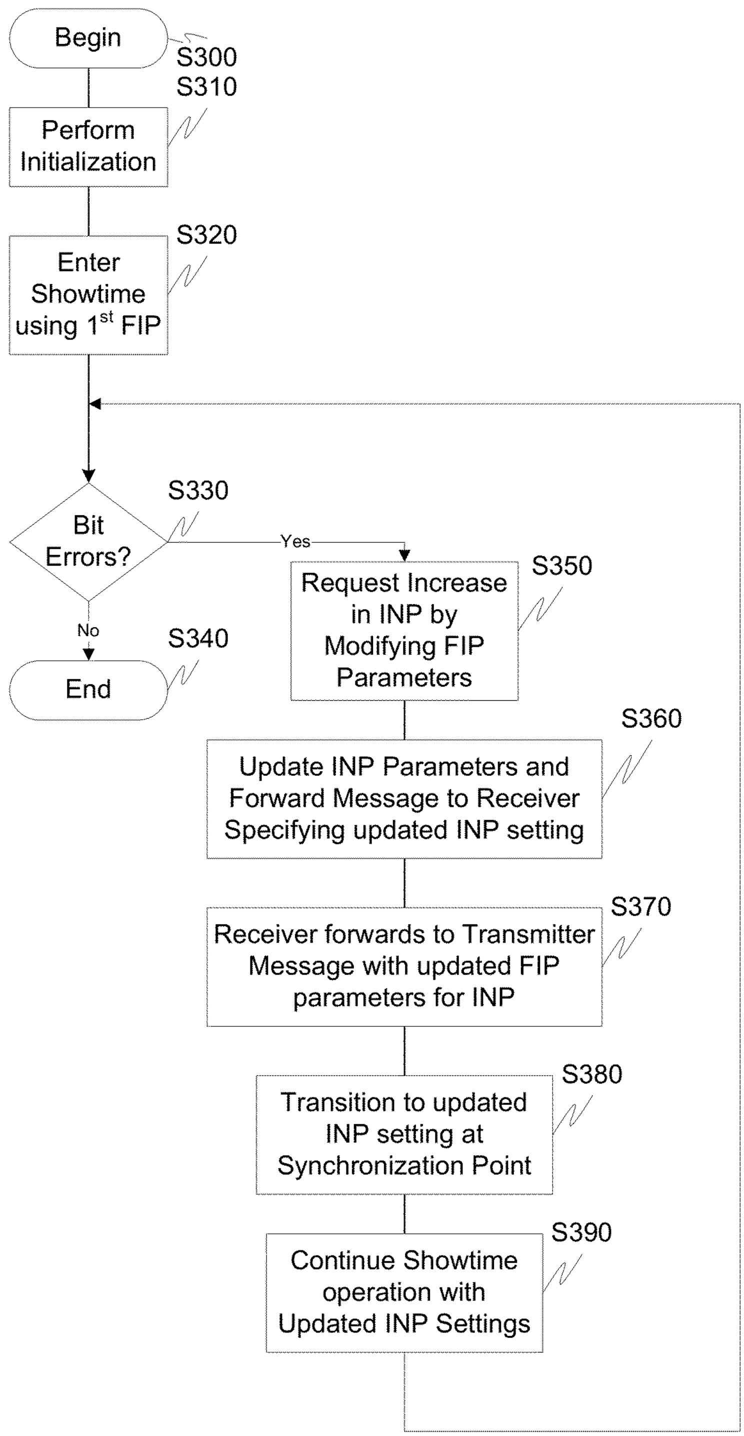
Begin
S300
S310
Perform Initialization
S320
Enter Showtime using 1st FIP
S330
Bit Errors?
Yes
No
S340
End
S350
Request Increase in INP by Modifying FIP Parameters
S360
Update INP Parameters and Forward Message to Receiver Specifying updated INP setting
S370
Receiver forwards to Transmitter Message with updated FIP parameters for INP
S380
Transition to updated INP setting at Synchronization Point
S390
Continue Showtime operation with Updated INP Settings

Fig. 3

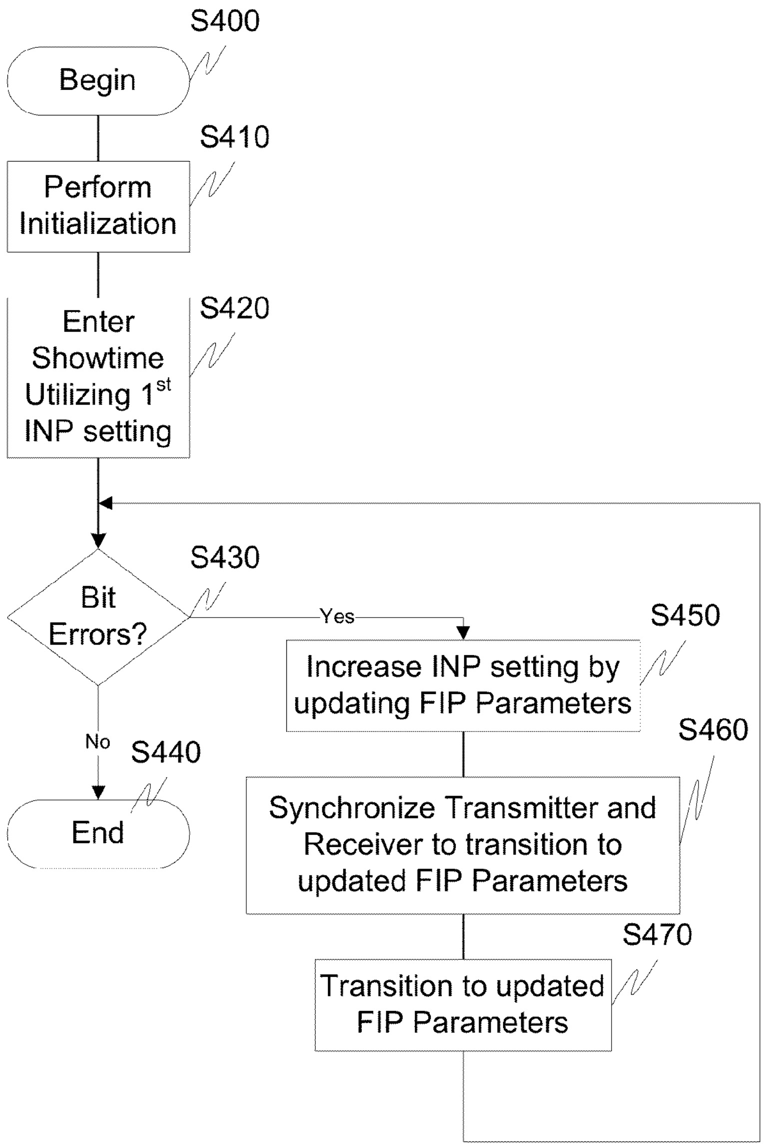
S400
Begin
S410
Perform Initialization
S420
Enter Showtime Utilizing 1st INP setting
S430
Bit Errors?
Yes
S450
Increase INP setting by updating FIP Parameters
No
S440
End
S460
Synchronize Transmitter and Receiver to transition to updated FIP Parameters
S470
Transition to updated FIP Parameters

Fig. 4

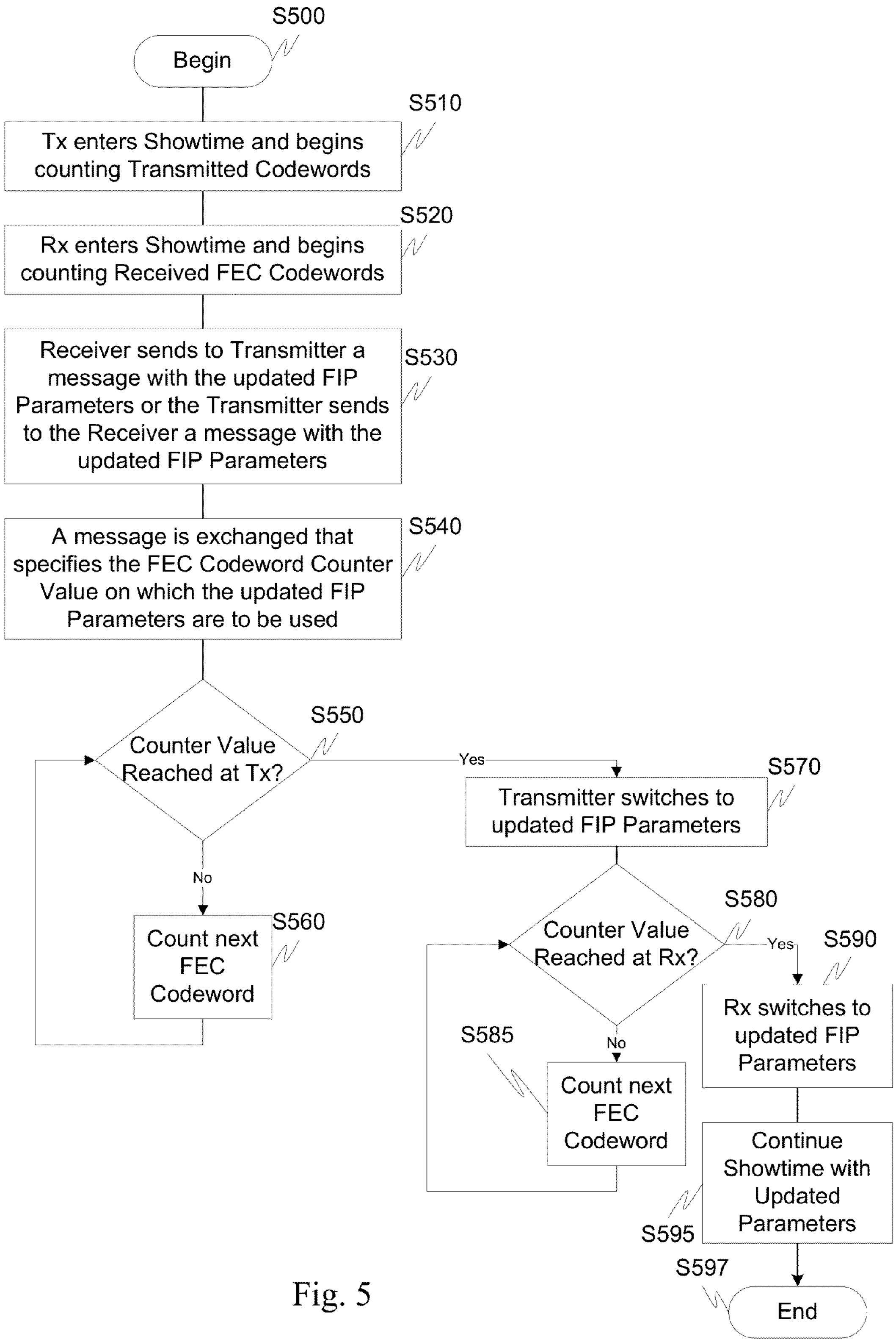
S500
Begin
S510
Tx enters Showtime and begins counting Transmitted Codewords
S520
Rx enters Showtime and begins counting Received FEC Codewords
S530
Receiver sends to Transmitter a message with the updated FIP Parameters or the Transmitter sends to the Receiver a message with the updated FIP Parameters
S540
A message is exchanged that specifies the FEC Codeword Counter Value on which the updated FIP Parameters are to be used
S550
Counter Value Reached at Tx?
Yes
No
S560
Count next FEC Codeword
S570
Transmitter switches to updated FIP Parameters
S580
Counter Value Reached at Rx?
Yes
No
S585
Count next FEC Codeword
S590
Rx switches to updated FIP Parameters
S595
Continue Showtime with Updated Parameters
S597
End

Fig. 5

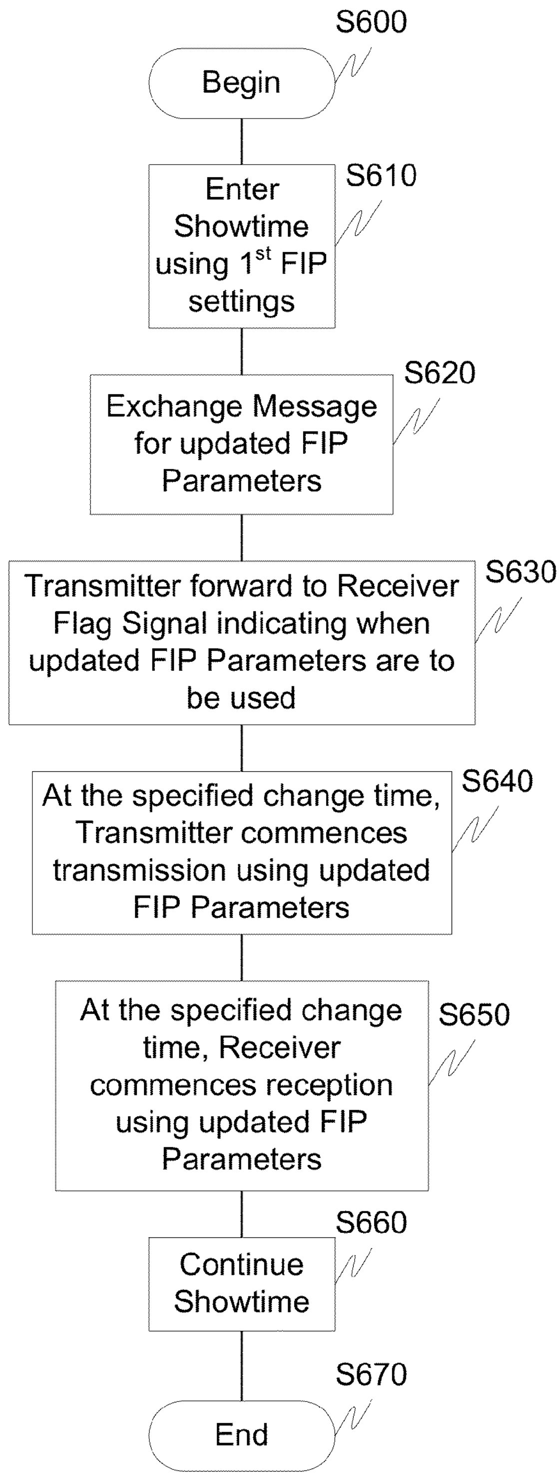
S600
Begin
S610
Enter Showtime using 1^{st} FIP settings
S620
Exchange Message for updated FIP Parameters
S630
Transmitter forward to Receiver Flag Signal indicating when updated FIP Parameters are to be used
S640
At the specified change time, Transmitter commences transmission using updated FIP Parameters
S650
At the specified change time, Receiver commences reception using updated FIP Parameters
S660
Continue Showtime
S670
End

Fig. 6

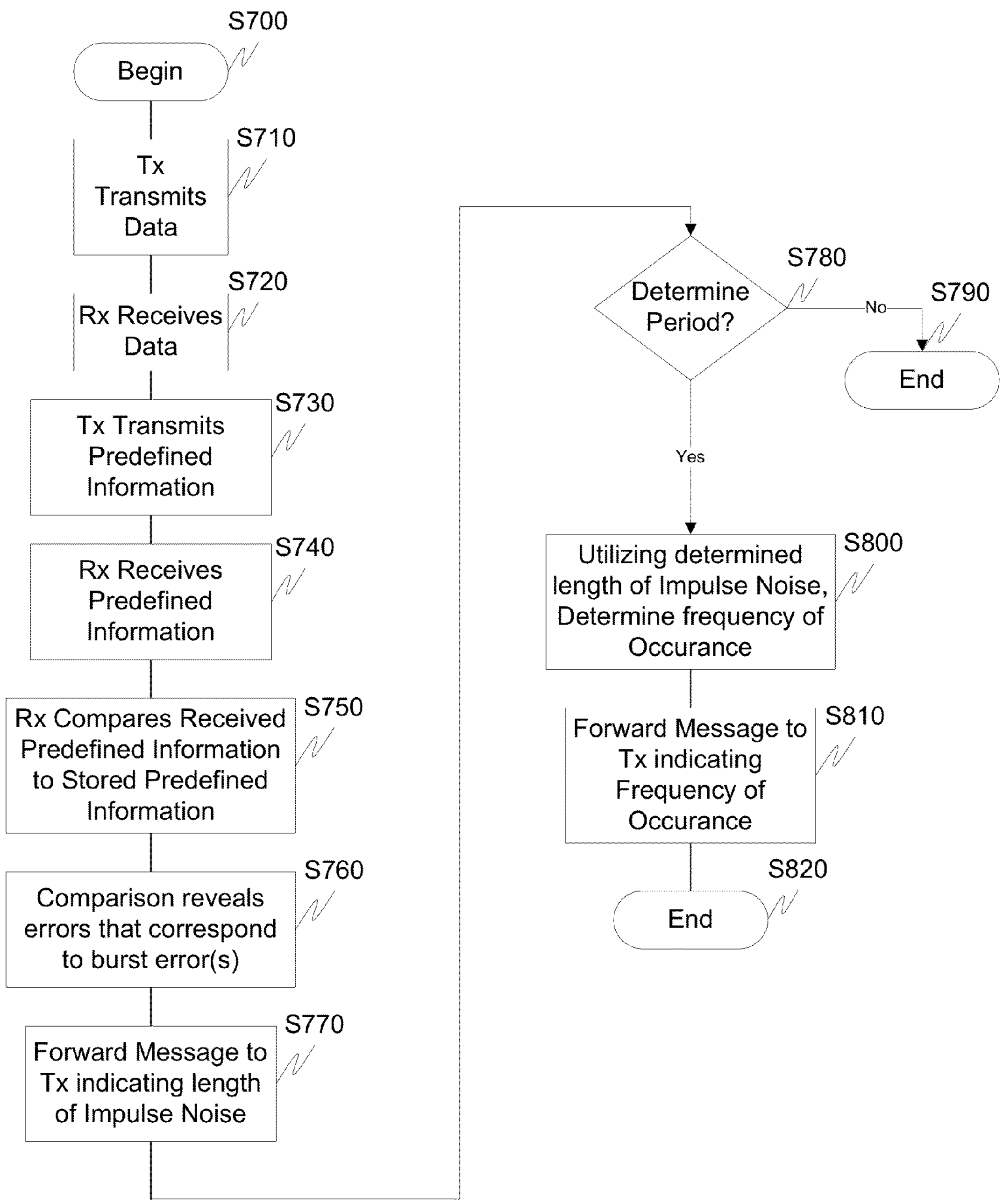
S700
Begin
S710
Tx Transmits Data
S720
Rx Receives Data
S730
Tx Transmits Predefined Information
S740
Rx Receives Predefined Information
S750
Rx Compares Received Predefined Information to Stored Predefined Information
S760
Comparison reveals errors that correspond to burst error(s)
S770
Forward Message to Tx indicating length of Impulse Noise
S780
Determine Period?
No
S790
End
Yes
S800
Utilizing determined length of Impulse Noise, Determine frequency of Occurance
S810
Forward Message to Tx indicating Frequency of Occurance
S820
End

Fig. 7

IMPULSE NOISE MANAGEMENT

RELATED APPLICATION DATA

This application is a Continuation of U.S. application Ser. No. 12/769,193 filed Apr. 28, 2010, now U.S. Pat. No. 8,462,835, which is a Continuation of U.S. application Ser. No. 10/597,482, filed Jul. 27, 2006, now abandoned, which is a national stage application under 35 U.S.C. 371 of PCT Application No. PCT/US2005/006842, filed Mar. 3, 2005, which claims the benefit of and priority under 35 U.S.C. §119(e) to U.S. Provisional Application No.: 60/549,804, entitled "On-Line Impulse Noise Protection (INP) Adaptation," filed Mar. 3, 2004, and U.S. Provisional Application No.: 60/555,982, entitled "Impulse Noise Protection (INP) Training," filed Mar. 24, 2004, each of which are incorporated herein by reference in their entirety.

BACKGROUND

1. Field of the Invention

This invention generally relates to communication systems. In particular, an exemplary aspect of this invention relates to impulse noise protection adaptation. Another exemplary aspect of this invention relates to impulse noise length and period determination and use thereof for impulse noise protection adaptation.

2. Description of Related Art

Communications systems often operate in environments that produce impulse noise. Impulse noise is a short-term burst of noise that is higher than the normal noise that typically exists in a communication channel. For example, DSL systems operate on telephone lines and experience impulse noise from many external sources including telephones, AM radio, HAM radio, other DSL services on the same line or in the same bundle, other equipment in the home, etc. It is standard practice for communications systems to use interleaving in combination with Forward Error Correction (FEC) to correct the errors caused by impulse noise. Standard initialization procedures in ADSL and VDSL systems are designed to optimize performance (data rate/reach) in the presence "stationary" crosstalk or noise. Impulse noise protection is handled with interleaving and FEC, but the current xDSL procedure at least does not provide specific states to enable training for the selection of the appropriate interleaving and FEC parameters.

An exemplary problem associated with traditional communication systems is that they use traditional Signal to Noise Ratio (SNR) measurement techniques to determine the SNR of the channel. These traditional techniques assume that the noise is stationary and does not contain non-stationary components such as impulse noise. The most common method for measuring the SNR is to calculate the mean-square error of the received signal based on a known transmitted signal, which is described in the ADSL series of ITU G.992.x standards and the VDSL series of ITU G.993.x standards, which are incorporated herein by reference in their entirety. These traditional methods for measuring SNR do not correctly measure the impact of impulse noise and do not have the noise capability to determine how the system should be configured to handle impulse noise.

There has been proposed that there is a need in ADSL and VDSL systems to provide robust error-free performance in the presence of high, real-world impulse noise. A specific proposal recommends that the standard impulse noise protection (INP) values are extended to values of 4, 8, 16 and 32 in order to handle high levels of impulse noise. Impulse noise

protection is defined in the ADSL2 Standard G.992.3, which is incorporated herein by reference in its entirety, as the number of impulse noise corrupted DMT symbols that can be corrected by the FEC and interleaving configuration. Specifically, G.992.3 defines the following variables:

INP=½*(S*D)*R/N

S=8*N/L

Latency (or delay)=S*D/4

Line Rate (in kbps)=L*4

where N is the codeword size in bytes, R is the number of parity (or redundancy) bytes in a codeword, D is the interleaver depth in number of codewords, and L is the number of bits in a DMT symbol.

If K is the number of information bytes in a codeword then:

$$N=K+R$$

and the user data rate is approximately equal to:

$$L*4*K/N.$$

In general, DSL systems (such as the one defined in ADSL G.992.x or VDSL G.993.x) use the FEC and Interleaving Parameters (FIP) characterized by the set of parameters (N, K, R, D). Using these parameters, the Burst Error Correction Capability (BECC) in bytes can be simply calculated as:

$$\text{BECC}=D*R/2 \text{ bytes}$$

where BECC is defined as the number of consecutive byte errors that can be corrected by the receiver. Note that if the receiver uses more intelligent decoding schemes, such as erasure detection, it is possible to correct even more than D*R/2 bytes. It also follows from above that INP=BECC/L.

The proposal further recommends that the higher INP values are achieved by increasing the amount of FEC redundancy while keeping the same system latency and the same interleaver memory at the expense of user data rate or excess margin. Since, on phone lines without excess margin, there is clearly a trade-off between high impulse noise protection values and user data rate, it would be advantageous to try to maximize the user data rate by finding the minimum impulse noise protection value that can provide adequate impulse noise protection. The current technique includes the steps of an operator, or service provider, configuring the ADSL connection with a specific noise protection value, the ADSL connection is initialized and the transceivers enter into steady state data transmission (i.e., Showtime), and if the connection is stable, i.e., error-free, then the service is acceptable and the process ends. If there are bit errors, then the process is repeated with the operator, or service provider, configuring the ADSL connection with another specific INP value.

One exemplary problem with this approach is that it is time consuming and can result in sub-optimum user rates. This is illustrated with reference to the following examples:

EXAMPLE 1

Assume that for a particular DSL connection there is high impulse noise and the required INP is 8. As a result, if the service provider uses a first INP configuration of 2, the DSL connection will not be error free. Therefore, the service provider needs to configure a higher INP value and reinitialize the connection. If a value of 4 is used as a second INP value, it still will not provide adequate impulse noise protection and bit errors will occur. Again, the service provider will need to configure a higher INP value until the correct value of 8 is configured. Clearly, the connection needs to be re-initialized every time there is a new INP configuration chosen and this trial and error technique proves to be very time consuming

EXAMPLE 2

Assume that for a particular DSL connection there is high-impulse noise and the required INP is 4. As a result, if the service provider uses a first INP configuration of 2, the connection will not be error free. Therefore, the service provider needs to configure a higher INP value and reinitialize the connection. In order to save time and not go through the number of initializations has occurred in Example 1, the service provider simply configures the system to the maximum INP value of 32. Obviously, there will be no bit errors with INP=32 since this connection needs only an INP value of 4. As a result, user data throughput is greatly degraded since the additional FEC redundancy will be three times higher than what is actually needed. For example, if the INP of 4 requires 10 percent FEC redundancy, an INP of 32 requires 40 percent FEC redundancy which results in a 30 percent decrease in user data rate.

SUMMARY

In additional to the above drawbacks, the related systems do not have the ability to actually measure the length or repetition period of impulse noise which can both be used to determine an appropriate impulse noise protection setting.

Exemplary aspects of this invention relate to determining the impact of impulse noise on a communication system and the capability to determine how the system should be configured to handle the impulse noise event.

An exemplary aspect of this invention determines the impact of impulse noise by transmitting and receiving using a plurality of different FEC and interleaving parameter settings. For each FEC and Interleaving Parameter (FIP) setting, the received signal quality is determined by, for example, detecting if there are bit errors after the receiver performs the FEC decoding and deinterleaving. Based on this, the appropriate FIP setting is selected and used for transmission and reception.

A plurality of FIP settings can be used for transmission and reception. In accordance with one particular aspect of this invention, the system can transition from one FIP setting to another FIP setting without going through the startup initialization procedure such as the startup initialization sequence utilized in traditional xDSL systems. For example, an xDSL system that implements the systems and methods described herein could start using an FIP setting of (N=255, K=247, R=8, D=64) and then transition to an FIP setting of (N=255, K=239, R=16, D=64) without re-executing the startup initialization procedure.

Knowing that the first FIP setting has a BECC=256 bytes and the second setting has a BECC=512 bytes, means, for example, that the second setting can correct an impulse that causes twice as many bit errors as the first FIP setting. On the other hand, the first FIP setting has less FEC parity (overhead) which results in a higher information (net) data rate for the user during Showtime. This is also illustrated by the fact that K, the number of information bytes per code word, is higher for the first FIP setting. For each of the FIP settings, the receiver detects whether there are bit errors after the decoding and deinterleaving process. This detection can be done by, for example, by performing a cyclic redundancy check (CRC) after the decoding and deinterleaving process is complete as defined in the ITU standard G.992.x. In general, the CRC is a well-known method for detecting bit errors. Since impulse noise occurs at random times, this system can operate using a particular FIP setting for a period of time that is sufficient to encounter the impulse noise. In the simple example illustrated above, only the K and R values were modified. It should be appreciated however, that the systems and methods of this invention are not limited thereto but rather can be extended to include the modification of any one or more FIP parameter(s).

The process of determining the impact of impulse noise by transmitting and receiving using a plurality of FIP settings can be done while in steady-state transmission, i.e., Showtime for DSL systems, when user information bits are being transmitted.

The process of determining the impact of impulse noise by transmitting and receiving using a plurality of FIP settings, can be done during a special impulse noise training period during which the system is not actually transmitting user data. In accordance with this exemplary aspect of the invention, the standard xDSL procedure is modified to include the capability of measuring the effectiveness of a chosen impulse noise protection (INP) setting during initialization and having receiver-controlled updates of transmission parameters that control the INP setting, e.g., FEC parameters and interleaving parameters, during initialization.

A new initialization state is included in the xDSL initialization procedure that provides the capability to measure the effectiveness of the current INP settings. The new initialization state will be referred to as the INPTraining state and it can, for example, follow the exchange of the Showtime transmission parameters such as the bi/gi table, trellis coding, tone reordering, FEC/interleaving parameters, etc. It is important to note that steady-state transmission during which user information is transmitted is known as "Showtime" in XDSL systems and that standard ITU G.992.3 ADSL systems and ITU VDSL G.993.3 systems include an exchange phase in initialization during which the Showtime parameters are exchanged, see, for example, G.992.3 and G.993.1.

During this INPTraining state, the transceivers transmit and receive using at least one of the standard Showtime functions using the Showtime parameters, e.g., bi/gi table, FEC/interleaving parameters, etc., that are exchanged during the previous exchange phase. These functions include at least one of Showtime PMD functions, e.g., bi/gi table, trellis coding, tone reordering, etc., PMS-TC functions, such as framing and FEC/interleaving, and TPS-TC functions. During the INPTraining state, the TPS-TC can transmit idle ATM cells, HDLC flags, or 64/65 idle packets depending on the TPS-TC type.

At the receiver, the CRC, the FEC, the TPS-TC error detection capabilities, and other receiver functions can be used to determine whether the INP setting is adequate for the current impulse noise conditions on the line. The receiver can also use these receiver functions to automatically and dynamically determine what the correct INP setting should be. If the current INP setting is not adequate, a new set of Showtime transmission parameters can be exchanged and the process repeated by reentering the INPTraining state using the newly exchanged Showtime transmission parameters. If, on the other hand, the current INP setting is adequate, the transceivers can enter into Showtime.

An exemplary advantage associated with this aspect of the invention is that the receiver can measure the effectiveness of the current INP setting and can make updates to the INP setting based on these measurements during initialization. This is important because the receiver generally has the most knowledge of channel conditions, receiver functionality, and processing capability. In general, the receiver has the best capability to make the necessary trade offs between data rate, latency, excess margin, FEC redundancy, coding gain, and the like. Current related ADSL systems have the operator servicing lines that are experiencing high impulse noise by trying

various INP settings in an attempt to find an error-free operating mode. However, since the operator is not capable of making the receiver trade offs stated above, such as the data rate, latency, and the like, the process will often lead to a sub-optimum result.

Another exemplary aspect of this invention relates to determining the length and/or repetition period of impulse noise events in order to select an appropriate INP setting.

For example, the transceivers can transmit and receive using at least one of the standard Showtime functions and parameters such as the bi/gi table, FEC/interleaving parameters, and the like. These functions include the Showtime PMD functions, such as the bi/gi table, trellis coding, tone reordering and the like, PMS-TC functions (such as framing and FEC interleaving) and TPS-TC functions. The TPS-TC may transmit ATM cells, HDLC packets, or 64/65 packets depending on, for example, the TPS-TC type.

At the receiver, the CRC, the FEC, the TPS-TC error detection capabilities, and other receiver functions can be used to determine the length and the period of the impulse noise and whether an INP setting is adequate for the current impulse noise conditions on the line.

For example, the receiver can determine the length of an impulse noise event. The length of impulse noise events can be determined in a number of ways. For example, they can be determined as:

1. The length of the impulse in time, e.g., how many microseconds the impulse power is above a specific noise level (e.g. the above the stationary channel noise)
2. The number received bits that are affected by the impulse noise, e.g. how many received bits are in error in a specific sliding time window or how many consecutive bits are in error in a stream
3. The number of received ATM cells that are affected by the impulse noise, e.g. how many ATM cells contain bits that are in error. This may be detected using a standard ATM HEC, which is a CRC that covers the ATM header bits. Alternatively this may detected by checking the ATM payload bits by, for example, transmitting a predefined bit pattern that is known by the receiver
4. The number of 64/65 packets that are affected by the impulse noise, e.g. how many packets contain bits that are in error (the 64/65 CRC can be used for this)
5. The number of received DMT symbols that are affected by the impulse noise, e.g., the measured noise in a DMT symbol is above a predefined threshold which results in most (if not all) the bits in that DMT symbol being in error
6. The number of FEC codewords that are affected by the impulse noise, e.g., how many codewords have an uncorrectable number of bit errors, i.e., the number of bit errors in a codeword exceeds the number of bit that are correctable by the FEC code

In accordance with one exemplary aspect, the FEC correction capability and interleaving is turned off when trying to determine the length of the impulse noise. For example, the FEC may be configured so that there are no parity bits (i.e., R=0) or the FEC may be disabled altogether (i.e., no codewords are sent). Additionally, for example, the interleaving may be disabled by setting the interleaver depth to 1 (i.e., D=1). Disabling the FEC and interleaving is beneficial when trying to determine the length of the impulse noise based on affected bits, affected code words and/or affected packets. This is true since when the FEC/interleaving is enabled, the impulse noise event will be spread over a large time period and it will be more difficult to determine the length of the original impulse.

The exemplary techniques discussed herein that are used to determine the length of the impulse event can also be extended to determining the repetition period of the impulse noise event, i.e., how often is impulse noise occurring. The repetition period can be important, because the period has an effect on the FIP setting that is used. For example, if the interleaving spreads an impulse noise event over a period of time that exceeds the impulse period, then the interleaver could combine multiple impulse noise events together. As a result, the FEC correction capability may have to be increased. In order to determine the repetition period of an impulse noise event, the receiver can first detect the impulse noise event as discussed above and then determine how often the impulse noise events occur. For example, periodic impulse noise due to AC power lines occurs at a 120 Hz reception rate, or approximately every 8 ms.

In the case where the impulse noise does not have a fixed length, i.e., where the impulse noise varies over time, the receiver can attempt to determine the maximum impulse noise length. This involves measuring several impulse noise events and determining the impulse noise event with the maximum length. In practice, it is likely that the impulse noise will have varying length and it is important that the FEC/interleaving settings be configured to handle the maximum, e.g., worst case, impulse noise event.

Once the receiver determines the (maximum) length of the impulse noise and/or the repetition period of the impulse noise, this information can be sent to the transmitting modem. In particular, when the receiving modem, such as a Customer-Premises (CPE) modem determines the (maximum) length of the impulse and/or the repetition period of the impulse, the CPE modem could send the information to a Central Office (CO) modem in a message. The length of the impulse length event can be defined and specified in the message in terms of time, received bits, ATM cells, 64/65 packets, DMT symbols, or the like. The CO modem could then, for example, provide this information to the CO-MIB, which is the management interface that is used by the operator or service provider to configure the modems. For example, based on this information, the operator may configure the modems to a different INP value, data rate, latency, or the like. This process could also be automated such that the message received by the CO modem allows automatic reconfiguration to adjust, for example, INP values, data rate, latency, or the like.

The exemplary aspects of the invention that are used to determine the length and/or repetition period of the impulse noise event can be performed during initialization and/or Showtime. During initialization, the method can perform during an INPTraining state such as the one described herein. During Showtime the method can be performed, for example, during a Showtime INP adaptation phase as described herein.

These and other features and advantages of this invention are described in, or are apparent from, the following description of the embodiments.

BRIEF DESCRIPTION OF THE DRAWINGS

The features of this invention will be described in relation to the figures, wherein:

FIG. **1** is a functional block diagram illustrating an exemplary impulse noise adaptation system according to this invention;

FIG. **2** illustrates an exemplary initialization state machine that includes an INPTraining state according to this invention;

FIG. **3** is a flowchart outlining an exemplary method for impulse noise protection adaptation during Showtime according to this invention;

FIG. **4** is a flowchart illustrating an exemplary receiver optimized method for impulse noise protection adaptation according to this invention;

FIG. **5** is a flowchart illustrating an exemplary method for forward error correction synchronization according to this invention;

FIG. **6** is a flowchart illustrating an exemplary method for flag signal synchronization according to this invention; and

FIG. **7** is a flowchart illustrating an exemplary method for determining impulse noise length and period according to this invention.

DETAILED DESCRIPTION

The exemplary embodiments of this invention will be described in relation to adapting impulse noise parameters as well as measuring impulse noise length and repetition period within an xDSL environment. However, it should be appreciated that, in general, the systems and methods of this invention can be applied and will work equally well with any type of communication system in any environment.

The exemplary systems and methods of this invention will be described in relation to xDSL modems and associated communication hardware, software, and communication channels. However, to avoid unnecessarily obscuring the present invention, the following description omits well-known structures and devices that may be shown in block diagram form or otherwise summarized.

For purposes of explanation, numerous details are set forth in order to provide a thorough understanding of the present invention. It should be appreciated however, that the present invention may be practiced in a variety of ways beyond the specific details set forth herein. For example, the systems and methods of this invention can generally be applied to any type of system within any environment in which the impulse noise length and/or period is desired, or for which impulse noise adaptation is desired.

Furthermore, while the exemplary embodiments illustrated herein show the various components of the system collocated in specific locations, it is to be appreciated that the various components of the system can be located or relocated at distant portions of a distributed network, such as a telecommunications network and/or the Internet, or within a dedicated secure, unsecured and/or encrypted system. Thus, it should be appreciated that the components of the system can be combined into one or more devices, such as a modem, or collocated on a particular node of a distributed network, such as a telecommunications network. As will be appreciated from the following description, and for reasons of computational efficiency, the components of the system can be arranged at any location within a distributed network without effecting the operation of the system. For example, the various components can be located in a central office (CO or ATU-C) modem, a customer premises modem (CPE or ATU-R), or some combination thereof. Similarly, the functionality of the system could be distributed between one or more of the modems and an associated computing device.

Furthermore, it should be appreciated that the various links, including communications link **20**, connecting the elements can be wired or wireless links, or any combination thereof or any other known or later-developed element(s) that is capable of supplying and/or communicating data to and from the connected elements. The term module as used herein can refer to any known or later-developed hardware, software, or combination of hardware and software that is capable of performing the functionality associated with that element. Furthermore, as used herein, the term "transmitter" has the same meaning as the term "transmitting modem" or "transmitting transceiver" and the term "receiver" has the same meaning as "receiving modem" or "receiving transceiver."

FIG. **1** illustrates an exemplary embodiment of an impulse noise adaptation system **100** according to an exemplary embodiment of this invention. In particular, the system **100** comprises a receiving modem **200** and a transmitting modem **300**. The exemplary receiving modem **200** comprises a decoder/deinterleaver **210**, a bit error rate (BER) detection module **220**, a forward error correction and interleaving parameter (FIP) module **230**, an impulse noise length determination module **240**, an impulse noise period determination module **250**, an impulse noise parameter management module **260**, an impulse noise protection adaptation module **270**, a synchronization module **280**, a message module **290** and a controller and memory (not shown) all interconnected by a link **5**. The exemplary transmitting modem **300** comprises a management module **310**, a synchronization module **320**, a message module **330**, and may optionally include an impulse noise parameter management module **340**.

In operation, the exemplary communication system adapts the impulse noise parameters on-line by operating using a series of different FIP settings. For each FIP setting, the system can dynamically determine if the appropriate amount of impulse noise protection is being provided. Based on these determinations, the system can select a particular FIP setting for regular, i.e., Showtime, operation. As will be discussed in greater detail hereinafter, this impulse noise protection adaptation can be performed during Showtime and/or during initialization.

Impulse noise protection adaptation during Showtime includes the following exemplary steps. First, the DSL system, i.e., the transmitting and receiving modems, completes regular initialization and commences transmission and reception using a first FIP setting. For DSL systems, this first FIP setting is selected by the receiver **200** and is based on the minimum/maximum data rate, maximum latency, and minimum INP parameters as configured by, for example, the service provider via the management module **310**. A detailed explanation of this procedure can be seen in G.992.3.

The system will use this first FIP setting for a period of time T**1**. During this period, and in conjunction with the BER detection module **220**, the receiver **200** detects if bit errors have occurred using this first FIP setting for the decoding and deinterleaving performed by the decoder/deinterleaver **210**. For example, the receiver **200** can use a CRC to detect bit errors. If there are no bit errors, then the current INP setting is adequate and there is no need to modify the INP setting and Showtime communication can continue as normal.

However, if bit errors have been detected by the BER detection module **220**, an adjustment to the INP setting may be appropriate. For example, a service provider, a user, or the system can choose an updated INP setting. Since bit errors have been detected, the INP setting could be increased by an on-line modification to the FIP parameters. Specifically, the transmitter **300**, and for example the management module **310**, in response to an indication from the receiver **200** that bit errors have been detected, can send a message to the receiver **200** to initiate a change in the INP settings. The receiver **200**, in cooperation with the message module **290**, the FIP module **230**, the impulse noise parameter management module **260** and impulse noise protection adaptation module **270**, can return a message to the transmitter **300** that specifies a newly determined FIP setting that satisfies the new INP requirement.

The transmitter **300** and receiver **200** can then transition to the new INP setting by starting to use the new FIP parameters

for transmission and reception, respectively, at a synchronized point in time. This synchronization can be done in accordance with a number of exemplary methods that are discussed hereinafter.

Upon synchronization, the system **100** continues communication using the updated FIP settings for a second period of time T**2**. If the receiver detects that bit errors are occurring using this updated FIP setting during the decoding and deinterleaving performed by the decoder/deinterleaver **210** the steps can be repeated with the selection of another updated INP setting. However, if there are no bit errors detected by the receiver **200** during the time period T**2**, then the new INP setting is adequate and there is no need for further change at which point the INP adaptation procedure can end.

This Showtime-based INP adaptation process can be repeated as many times as necessary until an INP setting is obtained that provides the required impulse noise immunity and/or bit error rate.

One advantage of this technique is that the transition between different FIP settings can be accomplished without reinitializing the transceivers using the lengthy standard initialization procedure that is typically used in ADSL and VDSL systems. In contrast, the transition can occur without the standard initialization since the transition between FIP settings can be synchronized between the transmitter and the receiver such that the receiver **200** can determine when to start FEC decoding using the new FIP settings for K and R. As will be discussed hereinafter, this transition can be synchronized using a number of different exemplary methods.

The transition can also be accomplished without synchronization in which case the receiver **200** will determine when the new FIP settings are used by some alternative means. For example, the receiver could determine when the new FIP settings are being used by FEC decoding using both the old and updated FIP settings and determining which one is currently being utilized by the transmitter by determining with which FIP setting the codeword is correct. Other receiver functions such as CRCs, ATM HEC errors and the like could also be used.

This impulse noise protection adaptation procedure can be performed during regular steady-state transmission, i.e., Showtime in ADSL, using actual user data or, for example, idle ATM cells. This methodology can also be performed during a special impulse noise training period during which the system is not actually transmitting user data. For example, this impulse noise training period could utilize transmitted predefined pseudorandom bit streams or, for example, idle ATM cells or HDLC flags for determining an appropriate INP value.

The length of the time periods T**1**, T**2**, . . . can be controlled by, for example, the receiving modem **200**. The receiving modem **200** could control the length of these time periods by, for example, sending a message that specifies how long the transmitter should transmit using a particular FIP setting. This length can be defined, for example, in terms of the number of DMT symbols or the number of FEC codewords. For example, a message could indicate that 200 DMT symbols should be sent for all FIP settings or 300 FEC code words should be sent for all FIP settings. Alternately, the message can indicate a different number of DMT symbols or FEC code words for each FIP setting. This technique could further be used to aid in determining an appropriate INP setting by forwarding a predetermined number of DMT symbols at a first FIP value, followed by a second number of DMT symbols at a second FIP value, and so on. For each of these sets of DMT symbols and associated FIP values, the receiver could determine the bit error rate, with the cooperation of the BER detection module **220**, and then forward a message, with the cooperation of the message module **290**, to the transmitter **300** specifying which FIP setting provided the appropriate impulse noise protection.

Similarly, the transmitting modem **300** and receiving modem **200** could cooperate such that an optimum impulse noise protection value is converged upon. For example, the transmitting modem **300**, with the cooperation of the message module **330**, could forward a message to the receiving modem **200** specifying a first INP setting. If there are no bit errors, a message could be sent via the message module **290** to the transmitting modem **300** indicating that there are no bit errors and a lower INP value could be attempted. The impulse noise parameter management module **260** and FIP module **230** could then determine an appropriate lower INP setting and forward it to the transmitter **300**. As discussed above, since a lower INP value is inversely proportional to the user data rate, it is advantageous to optimize the INP value based on detected errors. The transmitting modem **300**, in cooperation with the message module **330**, could then send another message to the receiving modem **200** indicating that the lower INP value will be transitioned to at which point the procedure repeats itself through detection of bit errors in cooperation with the bit error rate detection module **220** and the decoder/deinterleaver **210**. This reduction in the INP parameters can continue until bit errors are detected at which point a message is sent, with the cooperation of the message module **290**, to the transmitting modem **300** indicating a higher INP value is required since bit errors were detected. Then, for example, based on an evaluation of a convergence of bit errors vs. impulse noise protection parameters, an optimum INP value can be determined and transitioned to by the receiver **300** and transmitter **200**. This procedure could also be used to increase the INP parameters in a similar fashion.

The length of the time periods T**1**, T**2**, . . . can also be controlled by the transmitter **300**. For example, the transmitter **300** could control the length of these time periods, by, for example, sending a message to the receiver **200** that specifies the length of time the transmitter will transmit using a particular FIP setting. This length could be defined in terms of the number of DMT symbols or, for example, the number of FEC codewords. The exemplary message could indicate that, for example, 200 DMT symbols would be sent for all FIP settings or 300 FEC codewords will be sent for all FIP settings. The message could also indicate a different number of DMT symbols or FEC codewords for each FIP setting. In general, the message could contain any type of information relating to how long the transmitter will be using a specific FIP setting(s).

The length of the time periods T**1**, T**2** . . . can also be controlled by, for example, a service provider or a user. For example, the service provider could configure through the management module **310** a minimum time X=10 seconds to be used for testing each FIP setting. The service provider could use the knowledge of the nature of the impulse noise, such as how often impulse noise event occurs, to determine the times. However, and in general, the length of the time periods could be configured to be any length of time as appropriate.

As discussed above, when a determination is made that there are bit errors and an increase in the INP setting is desirable, the receiving modem **200** can determine the new INP settings. However, it should be appreciated that the transmitting modem **300**, in cooperation with the impulse noise parameter management module **340** could also determine updated INP settings. For example, either modem could con-

tinually adapt the INP values as discussed above until no bit errors that are the result of impulse noise are detected by the BER detection module **220**.

As alluded to above, the receiver and transmitter can synchronize the modification of the FEC and interleaving parameters such that the both the transmitter and receiver start using the parameters at the same instant in time. This synchronization can be based on, for example, a synchronization using FEC codeword counters or a flag signal.

For synchronization using FEC codeword counters, the receiver **200** and transmitter **300** can synchronize the change in cooperation with the sync modules **280** and **320**, by counting the FEC codewords from the beginning of Showtime and the transition would occur when a specific FEC codeword counter value that is known by both the transmitter **300** and the receiver **200** is reached. Prior to the transition point, the transmitter **300** or receiver **200** in cooperation with the message module **290** and message module **330**, would send a message indicating the FEC codeword count value on which the FIP parameters will be updated. For example, the transmitting modem **300** can enter into Showtime and the synchronization module **320** commences counting the number of transmitted FEC codewords. For example, the first codeword transmitted has a count value of 0, the second transmitted codeword has a count value of 1. . . . The counter can optionally be defined as having a finite length, for example, 0 to 1023 (10 bits) such that when the value of 1023 is reached, on the next FEC code word that is transmitted, the counter restarts at the value of 0.

Similarly, the receiving modem **200** upon entering Showtime starts a counter in cooperation with the synchronization module **280** that counts the number of received FEC codewords. The first received FEC codeword has a count value of 0, the second received FEC codeword to the count value of 1. . . . As with the synchronization module **320**, the synchronization module **280** can have a counter with a finite length, for example 0-1023, such that when the value of 1023 is reached, on the next FEC codeword the counter restarts at 0.

At some point, for example, based on detected bit errors, the user, a service provider, or the like, it is determined that an updated FIP setting is needed. The receiving modem **200** can send a message, via the message module **290**, to the transmitting modem **300**, and in particular the message module **330** and impulse noise parameter management module **340**. The updated FIP setting can alternatively be sent from the transmitting modem to the receiving modem.

The synchronization module **280** and message module **290** then cooperate to send a message to the transmitting modem **300** specifying the FEC codeword counter value on which the new FIP settings are to be used for transmission and reception. Alternatively, the transmitting modem, in cooperation with the message module **330**, can send a message to the receiving modem indicating the FEC codeword counter value on which the FIP settings are to be used for transmission and reception. For example, the message can indicate that when the code word counter equals 501, the new FIP setting will be used for transmission and reception.

When the transmitter FEC codeword counter equals the value indicated in the message, the synchronization module **220** instructs the transmitting modem **300** to transition to the new FIP settings. Similarly, when the synchronization module **280** in the receiving modem **200** counts the FEC codeword that equals the value indicated in the message, the receiving modem **200** transitions to using the new FIP settings for reception.

Synchronization can also be performed through the use of a flag signal (sync flag). For this exemplary embodiment, the receiving modem **200** and transmitting modem **300**, in cooperation with the synchronization module **280** and synchronization module **320**, synchronize the change in FIP settings using a flag or marker signal that is similar to that used in the ADSL2 G.992.3 ORL protocol. This protocol may be more desirable than using an FEC codeword counter because, for example, it has greater impulse noise immunity.

For synchronization using a flag signal, the receiver and transmitter would start using updated FEC and interleaving parameters on a pre-defined FEC codeword boundary following the sync flag. For example, while transmitting using a first INP setting, a determination is made by, for example, the BER detection module **220** that a new FIP setting is needed due to the presence of impulse noise on the line. This determination can be performed by the receiving modem, in cooperation with the FIP module **230**, the transmitting modem, a user, a service provider, or the like. The receiving modem **200**, in cooperation with the message module **290**, sends a message to the transmitting modem **300** indicating the new FIP settings to be used for transmission and reception. Alternatively, the transmitting modem, in cooperation with the message module **330**, prepares and sends a message to the receiving modem indicating the updated FIP settings to be used for transmission and reception.

The transmitting modem then sends a flag or marker signal to the receiving modem **200** indicating that the new FIP settings are to be used on a predetermined number of DMT symbols following the transmission of the flag or marker signal. For example, the flag signal could be an inverted sync symbol, or sync FLAG, as used in the ADSL2 G.992.3 OLR protocol. The transmitting modem **300** then starts using the new FIP settings for transmission on the predetermined number of DMT symbols following the transmission of the flag or marker signal. Similarly, the receiving modem starts using the new FIP settings for reception once the predetermined number of DMT symbols following the receipt of the flag or marker signal have been received.

EXAMPLE 1

On-Line INP Adaptation FIP Setting

This section describes an example of FIP settings for On-Line INP adaptation for DSL. In this example only the number of information bytes in a codeword (K) and the number of parity bytes in a codeword (R) are updated on-line. The Codeword Size (N) and Interleaver Depth (D) are not changed. This means that the latency (and interleaver memory size) and the line rate are not modified on-line. Since N=K+R this places restrictions on the allowed values for K and R.

1st Setting—{Approximate User Data Rate=3.968 Mbps, Line Rate=4.096 Mbps, N=128, K=124, R=4, S=1, D=64, Latency=16 msec, INP=1}

2nd Setting—{Approximate User Data Rate=3.840 Mbps, Line Rate=4.096 Mbps, N=128, K=120, R=8, S=1, D=64, Latency=16 msec, INP=2}

3nd Setting—{Approximate User Data Rate=3.584 Mbps, Line Rate=4.096 Mbps, N=128, K=112, R=16, S=1, D=64, Latency=16 msec, INP=4}

The On-line INP adaptation process is restricted to only modify the number of information bytes in a codeword (K) and the number of parity bytes in a codeword (R). The FEC Codeword Size (N=K+R) and Interleaver Depth (D) are not changed. This means that the latency (or interleaver memory size) and the line rate are not modified on-line. However the user data rate will change during the process since K is being modified. Also since the line rate and the FEC codeword size

are not modified, the S value does not change in the process. It is important to note that with these constraints, the On-line INP adaptation process can be done in as seamless manner (no bit errors and service interruption). This means provided that the modification if the FIP setting is restricted to K and R, the transition between FIP settings can be done in a seamless manner. This is the case because if the codeword size N and the interleaver depth D are not modified, the transition can happen without the problem of "interleaving memory flushing." Interleaver memory flushing is a well-known problem in which errors occur because interleaver and deinterleaver memory locations are overwritten due to on-line changes in the codeword size (N) and or interleaver depth (D).

EXAMPLE 2

On-Line INP Adaptation FIP Setting

This section describes an example of FIP settings for On-Line INP adaptation for DSL. In this example only the Codeword Size (N) and the number of parity bytes in a codeword (R) are updated on line. The number of information bytes in a codeword (K) and Interleaver Depth (D) are not changed and therefore the user data rate does not change. This means that the latency (and interleaver memory size) and the line rate are modified on-line. Since N=K+R this places restrictions on the allowed values for K and R.

1st Setting—{Approximate User Data Rate=3.968 Mbps, Line Rate=4.096 Mbps, N=128, K=124, R=4, S=1, D=64, Latency=16 msec, INP=1}

2nd Setting—{Approximate User Data Rate=3.968 Mbps, Line Rate=4.224 Mbps, N=132, K=124, R=8, S=1, D=64, Latency=16 msec, INP=2}

3nd Setting—{Approximate User Data Rate=3.968 Mbps, Line Rate=4.480 N=128, K=124, R=16, S=1, D=64, Latency=16 msec, INP=4}

In this example the Line Rate is modified on-line. For this reason it is necessary to also complete a rate adaptation process in order to complete this On-Line INP adaptation. A method for Seamless Rate Adaptation is described in U.S. Pat. No. 6,498,808.

While these examples restrict the changes to a subset of the FIP parameters, they can obviously be extended to cover any combination of the FIP parameters (N, K, R and D). For example, the value of D could also be modified in addition to the values of K, R and N. This could result in a change in the required interleaver memory and latency. In order to keep the memory and latency constant it is necessary to change the codeword size (N) accordingly when changing the interleaver depth (D). For example, if the interleaver depths is changed from D=64 to D=128, the Codeword size would have to be decreased by a factor of 2 so that overall latency is constant.

FIG. **2** illustrates an exemplary modified initialization state machine that includes the INPTraining state prior to Showtime. This state machine is based on the VDSL G.993.1 ITU standard. As defined in G.993.1, VTU-O is the VDSL Transceiver Unit at the Optical Network Unit (ONU) and VTU-R is the VDSL Transceiver Unit at the Remote terminal. The Initialization state machine in FIG. **2** is an example of how INPTraining states could be included in a modified VDSL initialization procedure. While exemplary FIG. **1** includes the INPTraining state at a specific time in the initialization sequence, the INPTraining state can be included at any time during initialization provided that it is preceded by a state during which Showtime parameters are exchanged between the transceivers.

O-P-INPTrain

During the O-P-INPTrain State the VTU-O transmitter transmits DMT Symbols using the standard PMD, PMS-TC and TPS-TC SHOWTIME functions with parameters exchanged during the previous Exchange Phase. During this state, the TPS-TC transmits idle ATM cells, HDLC flags or 64/65 idle packets. SOC is inactive during this state.

This state is used by the VTU-R to determine the correct DS INP setting based on Impulse Noise conditions on the line. For example, the downstream CRC, FEC error detection, TPS-TC error detection capabilities, and other receiver functions can be used to determine whether the INP setting is adequate. The receiver can also use these receiver functions to determine the correct INP setting.

If the VTU-R determines that the current INP setting is not adequate, the VTU-R transmits the R-P-ISYNCHRO2 signal to indicate the need to transition back to the Exchange Phase in order to exchange new transmission parameters.

If the VTU-R determines that the current INP setting is not adequate, the VTU-R transmits the R-P-SYNCHRO2 signal to indicate that it is OK to transition to Showtime with the current transmission parameters.

R-P-INPTrain

During the R-P-INPTrain State the VTU-R transmitter transmits DMT Symbols using the standard PMD, PMS-TC and TPS-TC Showtime functions with parameters exchanged during the previous Exchange Phase. During this state, the TPS-TC transmits idle ATM cells, HDLS flags or 64/65 idle packets. SOC is inactive during this state.

This state is used by the VTU-O to determine the correct US INP setting based on Impulse Noise conditions on the line. For example, the upstream (US) CRC, FEC error detection, TPS-TC error detection capabilities, and other receiver functions can be used to determine whether the INP setting is adequate. The receiver can also use these receiver functions to determine the correct INP setting.

If the VTU-O determines that the current INP setting is not adequate, the VTU-O transmits the O-P-ISYNCHRO2 signal to indicate the need to transition back to the Exchange Phase in order to exchange new transmission parameters.

If the VTU-O determines that the current INP setting is not adequate, the VTU-O transmits the O-P-SYNCHRO2 signal to indicate that it is OK to transition to Showtime with the current transmission parameters.

O-P-SYNCHRO2

The O-P-SYNCHRO2 is the same as defined in the current VDSL1. As in VDSL1, the VTU-O transmitter enters Showtime after transmitting the O-P-SYNCHRO2 signal.

But, if the VTU-R has not also entered into Showtime, the VTU-O waits for receipt of the R-P-SYNCHRO2 or R-P-ISYNCHRO2. If the VTU-O receives the R-P-SYNCHRO2 it continues in Showtime. If the VTU-O receives the R-P-ISYNCHRO2, the VTU-O transmitter transitions back to the beginning of the O-P-MEDLEY state.

R-P-SYNCHRO2

The O-P-SYNCHRO2 is the same as defined in the current VDSL1. As in VDSL1, the VTU-R transmitter enters Showtime after transmitting the R-P-SYNCHRO2 signal.

But, if the VTU-O has not also entered into Showtime, the VTU-R waits for receipt of the O-P-SYNCHRO2 or O-P-ISYNCHRO2. If the VTU-R receives the O-P-SYNCHRO2 it continues in Showtime. If the VTU-R receives the O-P-ISYNCHRO2, the VTU-R transmitter transitions back to the beginning of the R-P-MEDLEY state.

O-P-ISYNCHRO2

The O-P-ISYNCHRO2 is phase-inverted version of the O-P-SYNCHRO2, i.e., a subcarrier-by-subcarrier 180 degrees

phase reversal. The VTU-O transmitter transitions to the beginning of the O-P-MEDLEY state after transmitting the O-P-ISYNCHRO2 signal.

R-P-ISYNCHRO2

The R-P-ISYNCHRO2 is phase-inverted version of the R-P-SYNCHRO2, i.e., a subcarrier-by-subcarrier 180 degrees phase reversal. The VTU-R transmitter transitions to the beginning of the R-P-MEDLEY state after transmitting the R-P-ISYNCHRO2 signal

Exemplary Overview State Transition Rules Based on SYNCHRO2 and the ISYNCHRO2 signals:

If either the VTU-O or the VTU-R transmits the ISYNCHRO2 signal then both VTU-R and VTU-O transition back to beginning of the MEDLEY state.

If both the VTU-O and the VTU-R transmit the SYNCHRO2 signal then both the VTU-R and the VTU-O transition into Showtime.

There are several important points regarding this exemplary embodiment. First, the receiver can measure the efficiency of the plurality of INP settings without completing a new initialization procedure such as is used in ADSL and VDSL systems. Second, the length of the time of the O-P-INPTrain and R-P-INPTrain states can be controlled by the VTU-R and VTU-O respectively. In this way the receivers have adequate time to determine if the current INP setting is correct. In order to accomplish this, prior to entering the R-P-INPTrain state, the VTU-O can transmit a message to the VTU-R indicating the minimum length of the R-P-INPTrain state. Likewise, prior to entering the O-P-INPTrain state, the VTU-R can transmit a message to the VTU-O indicating the minimum length of the O-P-INPTrain state. For example, the message could indicate that a minimum of 20000 DMT symbols should be sent during the O-P-INPTrain state. Additionally, the length of the time of the O-P-INPTrain and R-P-INPTrain states could be set by the DSL service provider in order to make sure that the initialization does not take too long or because the service provider may have some knowledge of the statistics of the impulse noise which require setting the length of the INPTrain states. In this exemplary case a message could be sent from the VTU-O to the VTU-R indicating the minimum and/or maximum length of the O-P-INPTrain and/or the R-P-INPTrain states. For example, the message could indicate that a minimum of 20000 DMT symbols should be sent during the R-P-INPTrain state. Also, for example, the message could indicate that a maximum of 40000 DMT symbols should be sent during the R-P-INPTrain state.

The length of the time of the O-P-INPTrain and R-P-INPTrain states could also be controlled by the VTU-O and VTU-R respectively. In this case, the transmitters will have control of the state lengths. In order to accomplish this, prior to entering the R-P-INPTrain state, the VTU-R can transmit a message to the VTU-O indicating the minimum length of the R-P-INPTrain state. Likewise, prior to entering the O-P-INPTrain state, the VTU-O can transmit a message to the VTU-R indicating the minimum length of the O-P-INPTrain state. For example, the message could indicate that a minimum of 20000 DMT symbols should be sent during the O-P-INPTrain state.

As discussed above, another exemplary aspect of this invention relates to determining the length and/or repetition period of impulse noise events in order to select an appropriate INP setting.

The INP length can be determined using any one of a plurality of metrics. For example, the INP length can be determined based on one or more of:

1) The length of the impulse in time
2) The number of received bits that are affected by the impulse noise
3) The number of received ATM cells that are affected by the impulse noise
4) The number of 64/65 packets that are affected by the impulse noise
5) The number of received DMT symbols that are affected by the impulse noise
6) The number of FEC codewords that are affected by the impulse noise

As an example, assume that length of the impulse noise is 700 microseconds (Example 1). At a data rate of 1 Mbps this could correspond to an impulse noise length of 700 bits (Example 2) assuming that all impulse noise was high enough to affect all 700 bits in the 700 microsecond period. This also could correspond to a impulse noise length of 700/(53*8) =1.65 ATM cells since there are 53 bytes in a ATM cell (Example 3). This also could correspond to an impulse noise length of 700/(65*8)=1.35 64/65 packets since there are 65 bytes in a 64/65 packet (Example 4). This also could correspond to an impulse noise length of 700/(250)=2.8 DMT symbols (INP=2.8) assuming the DMT symbol rate is 4 kHz (250 microseconds DMT symbol length) (Example 5).

As an example, the determination of the INP length with the number of received bits that are affected by impulse noise can be performed in accordance with the following procedure. Initially, the transmitting modem **300** transmits data using at least one of the Showtime functions. In accordance with a first exemplary embodiment, at least the bit allocation table is used. In addition, or alternatively, the trellis coder, framer, and TPS-TC functions may also be used. Additionally, or alternatively still, the interleaving and FEC may also be used.

The receiving modem **200** receives data using at least one of the Showtime functions, such as the bit allocation table as discussed above. Similarly, the trellis coder, framer, TPS-TC and/or interleaving and FEC can be used.

The transmitter **300** then transmits a predefined bit pattern that is used for determining or measuring the impulse noise length based on detected erroneous bits. The receiving modem **200** compares, with the assistance of the impulse noise length determination module **240**, the predefined bit pattern to the received bit pattern in order to detect bit errors. Since impulse noise events typically cause a burst of errors in a bit stream, the receiving modem **200** determines the length of the impulse noise event by detecting and determining the length of the error burst.

Once the receiver **200** determines the length of the impulse noise, the receiving modem **200**, in cooperation with the message module **290**, sends a message to the transmitter **300** that indicates the determined length of the impulse noise.

In accordance with an exemplary embodiment, the FEC correction capability and interleaving is turned off when trying to determine the length of the impulse noise, e.g., R=0 and D=1. Disabling the FEC and interleaving is beneficial when trying to determine the length of the impulse noise based on affected bits, affected code words, or affected packets. This is the case because if the FEC/interleaving is enabled, the impulse event can be spread over a large time period which makes it more difficult to determine the length of the original impulse.

The exemplary methodology discussed above can also be used to determine the length of the impulse event using other metrics. For example, a predefined bit pattern could be used in the payload of the ATM cells or the 64/65 packets. Idle packets or cells, which carry a predefined pattern, could also be

used. In these cases, the receiver would compare the received data to the predefined data to determine the length of the impulse or to determine if an INP setting is adequate. Alternatively, the CRC of the ATM cell or 64/65 packet could be used to determine the length of the impulse noise. This would provide a courser measurement of the impulse noise length since it would be an integer number of packets or cells. In this example, a 700 bit impulse would cause CRC errors in two ATM cells so the length of the impulse noise would be two ATM cells, as opposed to the more precise measurement of 1.64 above. Likewise, in this example, a 700 bit impulse would cause CRC errors in two 64/65 packets so the length of the impulse noise would be two packets, as opposed to the more precise measure of 1.35 above.

Likewise, a predefined bit pattern could also be used to modulate the carriers in the DMT symbols so that the receiver would know what DMT symbols were transmitted. In this case, the receiver **200** would determine how many DMT symbols were corrupted by comparing the received signal with a known transmitted signal.

The exemplary techniques used to determine the length of the impulse noise event could also be extended to determine the repetition rate of the impulse noise event in cooperation with the impulse noise period determination module **250**. In order to determine the repetition period of an impulse noise event, the receiver **200** detects the impulse noise event as discussed above and then determines how often they occur. For example, periodic impulse noise due to AC power lines occur at 120 Hz reception rates or approximately every 8 ms. The receiver **200** could, for example, also store information about past impulse noise events and compare detected impulse noise events to historical events. The impulse noise period determination module **250** could then determine which events of a similar duration are occurring at what interval. This information could then be used in determining an appropriate INP setting.

Once the receiver **200** determines the (maximum) length of the impulse and/or the repetition period of the impulse, the information, with the cooperation of the message module **290** can be sent to the transmitting modem. For example, when the receiving modem **200** determines the (maximum) length of the impulse and/or the repetition period of the impulse, the impulse noise period determination module **250**, in cooperation with the message module **290** can forward information to the transmitting modem that quantifies this period. The transmitting modem **300** could provide this information to, for example, the management module **310**, that would allow, for example, an operator or service provider to configure the modems. For example, based on the period information contained in the message, the operator may configure the modems to different INP values, data rates, latency, or the like.

The receiver **200** could also test a specific INP setting by detecting how many received bits are errored in a specific time period. For example, if the specific INP setting enables the correction of 100 bits in an 8 msec. time period then the receiver **200** could detect how many bits are errored in a sliding 8 msec. window. If less than 100 bits are detected in error in the 8 ms sliding window, then the INP setting is adequate.

If more than 100 bits are detected in error in a sliding window, then the INP setting is not adequate, and the FEC/interleaving needs to be changed to provide more error correction capability.

Likewise, instead of using the received number of bits that are effected by the impulse noise, the receiver could test a specific INP setting by detecting how many received ATM cells, 64/65 packets, DMT symbols, and/or FEC corrections are affected in a specific time period.

FIG. **3** outlines an exemplary method for performing impulse noise protection adaptation during Showtime according to this invention. In particular, control begins in step **S300** and continues to step **S310**. In step **S310**, traditional DSL initialization occurs. Next, in step **S320**, Showtime is entered between the two modems using the first FIP setting that was determined during the initialization in step **S310**. Then, in step **S330**, a determination is made whether bit errors are occurring using the first FIP setting. If bit errors are not occurring, control continues to step **S340** where the control sequence ends. Otherwise, control jumps to step **S350**.

In step **S350**, a determination is made that an increase of the INP setting is required that requires modification of the FIP parameters. Next, in step **S360**, updated INP parameter is determined and a message forwarded to the receiver specifying the new INP setting. Then, in step **S370**, the receiver forwards to the transmitter updated FIP parameters for the new impulse noise protection requirements. Control then continues to step **S380**.

In step **S380**, the transmitter and receiver transition to using the updated INP parameters at a synchronization point. Next, in step **S390** Showtime operation continues. Control then continues back to step **S330**.

FIG. **4** outlines an exemplary method for receiver optimized impulse noise protection adaptation during Showtime. In particular, control begins in step **S400** and continues to step **S410**. In step **S410**, the DSL system completes regular initialization In particular, control begins in step **S400** and continues to step **S410**. In step **S410**, the DSL system completes startup initialization and continues into Showtime in step **S420** using a first FIP setting. Control then continues to step **S430** where a determination is made whether bit errors are occurring using the first FIP setting. If bit errors are not occurring, control continues to step **S440** where the control sequence ends.

Otherwise, control jumps to step **S450** where the receiver increases the impulse noise protection by modifying the FIP parameters. Next, in step **S460**, a synchronization point is determined between the transmitter and receiver, and when the synchronization point is reached both the transmitter and the receiver transition to the updated FIP setting in step **S470** and Showtime communications continue and control returns back to step **S430**.

FIG. **5** illustrates an exemplary method for synchronization of the modified FEC and interleaving parameters according to this invention. In particular, control begins in step **S500** and continues to step **S510**. In step **S510**, the transmitter enters Showtime and counts the number of transmitted FEC codewords. Next, in step **S520**, the receiver enters Showtime and counts the number of received FEC codewords. Then, in step **S530**, after a determination is made that an updated FIP setting is needed, the receiving modem sends a message to the transmitting modem indicating a new FIP setting that is to be used for transmitting and reception or, alternatively, the transmitting modem sends a message to the receiving modem indicating the new FIP setting to be used for transmission and reception. Control then continues to step **S540**.

In step **S540**, a message with the FEC codeword counter value on which the new FIP values are to be used is exchanged. Next, in step **S550**, a determination is made whether the counter value has been reached at the transmitter. If the counter value has not been reached, the next codeword is counted in step **S560** and control continues back to step **S550**.

Otherwise, if the counter value has been reached, control jumps to step S570. In step S570, the transmitter transitions to the new FIP setting. Next, in step S580, a determination is made whether the counter value has been reached at the receiving modem. If the counter value has not been reached, the next received FEC codeword is counted in step S585 and control continues back to step S580. However, when the counter value has been reached in the receiving modem, control jumps to step S590 where the receiving modem switches to the new FIP value. Control then continues to step S595 where the modems continue Showtime communication and control continues to step S597 where the control sequence ends.

FIG. 6 illustrates an exemplary method of synchronization using a flag signal according to this invention. In particular, control begins in step S600 and continues to step S610. In step S610, the modems enter Showtime using the first FIP parameters. Next, in step S620, a message is exchanged indicating the new FIP settings. Then, in step S630, the transmitter forwards to the receiver a flag signal indicating when the new FIP settings are to be used.

At step S640, and at a predefined change time following the transmission of the flag signal, the transmitter begins transmission using the new FIP parameters. Next, at step S650, at the predefined change time following the reception of the flag signal, the receiver commences reception utilizing the new FIP parameters. Control then continues to step S660 where Showtime communication continues with the control sequence ending at step S670.

FIG. 7 illustrates an exemplary method of impulse noise length and period determination. In particular, control begins in step S700 and continues to step S710. In step S710, the transmitter transmits data using at least one Showtime function. Next, in step S720, the receiver receives data using at least one Showtime function. Then, in step S730, the transmitter transmits predefined information to the receiver. Control then continues to step S740.

In step S740, the receiver receives the predefined information from the transmitter. Next, in step S750, the receiver compares the received predefined information to the predefined information and determines the differences (i.e., errors) between the two. Then, in step S760, and based on the detected errors, the length of the burst error is determined. Next, in step S770, a message is forwarded to the transmitter indicating the length of the impulse noise event. Control then continues to step S780 where a determination is made whether the period of the impulse noise event is also to be determined. If the period is not to be determined, control continues to step S790 where the control sequence ends. Otherwise, control jumps to step S800.

In step S800, and once the length of the impulse noise event is know, the receiver detects how often the impulse noise events occur. For example, historical information regarding the length and timing of previous impulse noise events can be stored in a memory (not shown). Then, a comparison can be made with the aid of a processor (not shown) to compare an impulse noise event to the historical information to determine the period of repetition (if any) of similar impulse noise events and, for example, a message indicating the period as well as the timing forwarded to, for example, another transceiver, the CO, or in general any destination as appropriate. Thus, in a similar manner, this information can be forwarded in step S810 to, for example, the transmitter in a message specifying the repetition frequency of the impulse noise event. Control then continues to step S820 where the control sequence ends.

The above-described system can be implemented on wired and/or wireless telecommunications device(s), such a modem, a multicarrier modem, a DSL modem, an ADSL modem, an xDSL modem, a VDSL modem, a linecard, test equipment, a multicarrier transceiver, a wired and/or wireless wide/local area network system, a satellite communication system, a modem equipped with diagnostic capabilities, or the like, or on a separate programmed general purpose computer having a communications device.

Additionally, the systems, methods and protocols of this invention can be implemented on a special purpose computer, a programmed microprocessor or microcontroller and peripheral integrated circuit element(s), an ASIC or other integrated circuit, a digital signal processor, a hard-wired electronic or logic circuit such as discrete element circuit, a programmable logic device such as PLD, PLA, FPGA, PAL, modem, transmitter/receiver, or the like. In general, any device capable of implementing a state machine that is in turn capable of implementing the methodology illustrated herein can be used to implement the various communication methods, protocols and techniques according to this invention.

Furthermore, the disclosed methods may be readily implemented in software using object or object-oriented software development environments that provide portable source code that can be used on a variety of computer or workstation platforms. Alternatively, the disclosed system may be implemented partially or fully in hardware using standard logic circuits or VLSI design. Whether software or hardware is used to implement the systems in accordance with this invention is dependent on the speed and/or efficiency requirements of the system, the particular function, and the particular software or hardware systems or microprocessor or microcomputer systems being utilized. The communication systems, methods and protocols illustrated herein however can be readily implemented in hardware and/or software, or any means, using any known or later developed systems or structures, devices and/or software by those of ordinary skill in the applicable art from the functional description provided herein and with a general basic knowledge of the computer and telecommunications arts.

Moreover, the disclosed methods may be readily implemented in software, that can be stored on a storage medium, and executed on programmed general-purpose computer, a special purpose computer, a microprocessor, or the like. In these instances, the systems and methods of this invention can be implemented as a program embedded on personal computer such as JAVA® or CGI script, as a resource residing on a server or computer workstation, as a routine embedded in a dedicated communication system or system component, or the like. The system can also be implemented by physically incorporating the system and/or method into a software and/or hardware system, such as the hardware and software systems of a communications transceiver.

It is therefore apparent that there has been provided, in accordance with the present invention, systems and methods for impulse noise adaptation. While this invention has been described in conjunction with a number of embodiments, it is evident that many alternatives, modifications and variations would be or are apparent to those of ordinary skill in the applicable arts. Accordingly, it is intended to embrace all such alternatives, modifications, equivalents and variations that are within the spirit and scope of this invention.

The invention claimed is:

1. A method, in a transceiver, comprising:

transmitting using a first interleaver parameter value;

transmitting a flag signal; and

changing to transmitting using a second interleaver parameter value that is different than the first interleaver parameter value,

wherein the second interleaver parameter value is used for transmission on a pre-defined forward error correction codeword boundary following transmission of the flag signal.

2. The method of claim **1**, wherein the flag signal is an inverted sync symbol.

3. The method of claim **1**, wherein the change in interleaver value does not cause bit errors or service interruption.

4. The method of claim **1**, wherein the change in interleaver value is associated with at least one of an impulse noise protection value, a data rate and a latency value.

5. The method of claim **1**, wherein the change in interleaver value is associated with a service provider configuration.

6. The method of claim **1**, further comprising transmitting the flag signal on a telephone line that experiences impulse noise from external sources including one or more of AM radio, HAM radio and AC power lines.

7. The method of claim **1**, wherein the transceiver is located in a linecard that includes a management interface that is usable by an operator or service provider to configure a service.

8. A device comprising:

an interleaver configured to interleave a plurality of bits; and

a transmitter portion coupled to the interleaver and configured to:

transmit using a first interleaver parameter value;

transmit a flag signal; and

change to transmitting using a second interleaver parameter value that is different than the first interleaver parameter value,

wherein the second interleaver parameter value is used for transmission on a pre-defined forward error correction codeword boundary following transmission of the flag signal.

9. The device of claim **8**, wherein the flag signal is an inverted sync symbol.

10. The device of claim **8**, wherein the change in interleaver value does not cause bit errors or service interruption.

11. The device of claim **8**, wherein the change in interleaver value is associated with at least one of an impulse noise protection value, a data rate and a latency value.

12. The device of claim **8**, wherein the change in interleaver value is associated with a service provider configuration.

13. The device of claim **8**, wherein the transmitter is further configured to transmit the flag signal on a telephone line that experiences impulse noise from external sources including one or more of AM radio, HAM radio and AC power lines.

14. The device of claim **8**, wherein the transmitter is located in a linecard that includes a management interface that is usable by an operator or service provider to configure a service.

15. The device of claim **8**, wherein the transmitter is located in a Customer Premises Equipment (CPE) that includes a management interface that is usable by an operator, a service provider or service user.

16. The device of claim **8**, wherein the transmitter includes at least one digital signal processor.

17. The device of claim **8**, wherein the transmitter includes at least one ASIC (Application-Specific Integrated Circuit).

* * * * *

IN THE UNITED STATES DISTRICT COURT
FOR THE DISTRICT OF DELAWARE

TQ DELTA, LLC, Plaintiff, v. COMCAST CABLE COMMUNICATIONS, LLC Defendant.	Civil Action No. 1:15-cv-00611-RGA
TQ DELTA, LLC, Plaintiff, v. COXCOM LLC and COX COMMUNICATIONS INC., Defendants.	Civil Action No. 1:15-cv-00612-RGA
TQ DELTA, LLC, Plaintiff, v. DIRECTV, LLC, Defendant.	Civil Action No. 1:15-cv-00613-RGA

TQ DELTA, LLC, Plaintiff, v. DISH NETWORK CORPORATION, DISH NETWORK LLC, DISH DBS CORPORATION, ECHOSTAR CORPORATION, and ECHOSTAR TECHNOLOGIES, LLC Defendants.	Civil Action No. 1:15-cv-00614-RGA
TQ DELTA, LLC, Plaintiff, v. TIME WARNER CABLE INC. and TIME WARNER CABLE ENTERPRISES LLC, Defendants.	Civil Action No. 1:15-cv-00615-RGA
TQ DELTA, LLC, Plaintiff, v. VERIZON SERVICES CORP., Defendant.	Civil Action No. 1:15-cv-00616-RGA

MEMORANDUM OPINION

Brian E. Farnan, Esq., FARNAN LLP, Wilmington, DE; Michael J. Farnan, Esq., FARNAN LLP, Wilmington, DE; Peter J. McAndrews, Esq. (argued), MCANDREWS, HELD & MALLOY, LTD., Chicago, IL; Thomas J. Wimbiscus, Esq., MCANDREWS, HELD & MALLOY, LTD., Chicago, IL; Scott P. McBride, Esq., MCANDREWS, HELD & MALLOY, LTD., Chicago, IL; Rajendra A. Chiplunkar, Esq., MCANDREWS, HELD & MALLOY, LTD., Chicago, IL; James P. Murphy, Esq., MCANDREWS, HELD & MALLOY, LTD., Chicago, IL.

Attorneys for Plaintiff

Jack B. Blumenfeld, Esq., MORRIS, NICHOLS, ARSHT & TUNNELL LLP, Wilmington, DE; Jennifer Ying, Esq., MORRIS, NICHOLS, ARSHT & TUNNELL LLP, Wilmington, DE; L. Norwood Jameson, Esq., DUANE MORRIS LLP, Atlanta, GA; Matthew C. Gaudet, Esq. (argued), DUANE MORRIS LLP, Atlanta, GA; Corey J. Manley, Esq., DUANE MORRIS LLP, Atlanta, GA; David C. Dotson, Esq., DUANE MORRIS LLP, Atlanta, GA; S. Neil Anderson, Esq., DUANE MORRIS LLP, Atlanta, GA; Alice E. Snedeker, Esq., DUANE MORRIS LLP, Atlanta, GA; John M. Baird, Esq., DUANE MORRIS LLP, Washington, DC.

Attorneys for Defendants Comcast Cable Communications LLC, CoxCom LLC, Cox Communications Inc., DIRECTV, LLC, Time Warner Cable Inc., and Time Warner Cable Enterprises LLC

Alex V. Chachkes, Esq. (argued), ORRICK, HERRINGTON & SUTCLIFFE LLP, New York, NY.

Attorney for Defendant DIRECTV, LLC.

Rodger D. Smith, II, Esq., MORRIS, NICHOLS, ARSHT & TUNNELL LLP, Wilmington, DE; Eleanor G. Tennyson, Esq., MORRIS, NICHOLS, ARSHT & TUNNELL LLP, Wilmington, DE; Heidi L. Keefe, Esq. (argued), COOLEY LLP, Palo Alto, CA; Stephen P. McBride, COOLEY LLP, Palo Alto, CA;.

Attorneys for Defendants Dish Network Corporation, Dish Network LLC, Dish DBS Corporation, Echostar Corporation and Echostar Technologies, LLC.

Benjamin J. Schladweiler, Esq., ROSS ARONSTAM & MORITZ, Wilmington, DE.

Attorney for Defendant Verizon Services Corp.

November 30, 2016

ANDREWS, U.S. DISTRICT JUDGE:

Presently before the Court is the issue of claim construction of multiple terms in U.S. Patent Nos. 8,718,158 ("the '158 patent"), 9,014,243 ("the '243 patent"), 8,611,404 ("the '404 patent"), 9,094,268 ("the '268 patent"), 7,835,430 ("the '430 patent"), and 8,238,412 ("the '412 patent"). The Court has considered the Parties' Joint Claim Construction Brief. (Civ. Act. No. 15-611-RGA, D.I. 144; Civ. Act. No. 15-612-RGA, D.I. 141; Civ. Act. No. 15-613-RGA, D.I. 141; Civ. Act. No. 15-614-RGA, D.I. 135; Civ. Act. No. 15-615-RGA, D.I. 141; Civ. Act. No. 15-616-RGA; D.I. 146).[1] The Court heard oral argument on October 18, 2016. (D.I. 158).

I. BACKGROUND

Plaintiff filed these actions on July 17, 2015, alleging infringement of eight patents. (D.I. 1). On July 14, 2016, Plaintiff dismissed two of these patents with prejudice. (D.I. 102). The parties divide the remaining contested patents into three groupings: the phase scrambling patents, the low power mode patents, and the diagnostic mode patents. The phase scrambling patents, which include the '158 and '243 patents, claim methods for reducing the peak to average power ratio of a multicarrier transmission system. The low power mode patents, which include the '404 and '268 patents, claim methods for causing a multicarrier communications system to enter a low power mode while storing state information for full power mode to enable a rapid start up without the need for reinitialization. The diagnostic mode patents, which include the '430 and '412 patents, claim both an apparatus and method for the reliable exchange of diagnostic and test information over a multicarrier communications system.

[1] Unless otherwise specifically noted, all references to the docket refer to Civil Action No. 15-611-RGA.

II. LEGAL STANDARD

"It is a bedrock principle of patent law that the claims of a patent define the invention to which the patentee is entitled the right to exclude." *Phillips v. AWH Corp.*, 415 F.3d 1303, 1312 (Fed. Cir. 2005) (en banc) (internal quotation marks omitted). "'[T]here is no magic formula or catechism for conducting claim construction.' Instead, the court is free to attach the appropriate weight to appropriate sources 'in light of the statutes and policies that inform patent law.'" *SoftView LLC v. Apple Inc.*, 2013 WL 4758195, at *1 (D. Del. Sept. 4, 2013) (quoting *Phillips*, 415 F.3d at 1324) (alteration in original). When construing patent claims, a court considers the literal language of the claim, the patent specification, and the prosecution history. *Markman v. Westview Instruments, Inc.*, 52 F.3d 967, 977–80 (Fed. Cir. 1995) (en banc), *aff'd*, 517 U.S. 370 (1996). Of these sources, "the specification is always highly relevant to the claim construction analysis. Usually, it is dispositive; it is the single best guide to the meaning of a disputed term." *Phillips*, 415 F.3d at 1315 (internal quotation marks omitted).

"[T]he words of a claim are generally given their ordinary and customary meaning. . . . [Which is] the meaning that the term would have to a person of ordinary skill in the art in question at the time of the invention, i.e., as of the effective filing date of the patent application." *Id.* at 1312–13 (citations and internal quotation marks omitted). "[T]he ordinary meaning of a claim term is its meaning to [an] ordinary artisan after reading the entire patent." *Id.* at 1321 (internal quotation marks omitted). "In some cases, the ordinary meaning of claim language as understood by a person of skill in the art may be readily apparent even to lay judges, and claim construction in such cases involves little more than the application of the widely accepted meaning of commonly understood words." *Id.* at 1314.

When a court relies solely upon the intrinsic evidence—the patent claims, the specification, and the prosecution history—the court's construction is a determination of law. *See Teva Pharm. USA, Inc. v. Sandoz, Inc.*, 135 S. Ct. 831, 841 (2015). The court may also make factual findings based upon consideration of extrinsic evidence, which "consists of all evidence external to the patent and prosecution history, including expert and inventor testimony, dictionaries, and learned treatises." *Phillips*, 415 F.3d at 1317–19 (internal quotation marks omitted). Extrinsic evidence may assist the court in understanding the underlying technology, the meaning of terms to one skilled in the art, and how the invention works. *Id.* Extrinsic evidence, however, is less reliable and less useful in claim construction than the patent and its prosecution history. *Id.*

"A claim construction is persuasive, not because it follows a certain rule, but because it defines terms in the context of the whole patent." *Renishaw PLC v. Marposs Societa' per Azioni*, 158 F.3d 1243, 1250 (Fed. Cir. 1998). It follows that "a claim interpretation that would exclude the inventor's device is rarely the correct interpretation." *Osram GMBH v. Int'l Trade Comm'n*, 505 F.3d 1351, 1358 (Fed. Cir. 2007) (citation and internal quotation marks omitted).

III. CONSTRUCTION OF DISPUTED TERMS

A. The Phase Scrambling Patents

The '158 patent is directed to a method for scrambling the phase characteristics of carrier signals in a multicarrier communications system. Claim 1 is representative and reads as follows:

> 1. In a *multicarrier* modulation system including a first *transceiver* in communication with a second *transceiver* using a transmission signal having a plurality of *carrier signals* for modulating a plurality of data bits, each *carrier signal* having a phase characteristic associated with at least one bit of the plurality of data bits, a method for *scrambling the phase characteristics of the carrier signals* comprising:
>
> transmitting the plurality of data bits from the first *transceiver* to the second *transceiver*;

> associating a *carrier signal* with a value determined independently of any bit of the plurality of data bits carried by the *carrier signal*, the value associated with the *carrier signal* determined by a pseudo-random number generator;
>
> *determining a phase shift for the carrier signal* at least based on the value associated with the *carrier signal*;
>
> modulating at least one bit of the plurality of data bits on the *carrier signal*;
>
> modulating the at least one bit on a second *carrier signal* of the plurality of *carrier signals*.

('158 patent, claim 1) (disputed terms italicized).

The '243 patent is also directed to a method for scrambling the phase characteristics of carrier signals in a multicarrier communications system. Claim 1 is representative and reads as follows:

> 1. A method, in a *multicarrier* communications *transceiver* comprising a *bit scrambler* followed by a *phase scrambler*, comprising:
>
> scrambling, using the *bit scrambler*, a plurality of input bits to generate a plurality of scrambled output bits, wherein at least one scrambled output bit is different than a corresponding input bit;
>
> scrambling, using the *phase scrambler*, a plurality of *carrier* phases associated with the plurality of scrambled output bits;
>
> transmitting at least one scrambled output bit on a first *carrier*; and
>
> transmitting the at least one scrambled output bit on a second *carrier*.

('243 patent, claim 1) (disputed terms italicized).

1. "carrier signal" and "carrier"

   a. *Plaintiff's proposed construction*: "plain meaning"

   b. *Defendants' proposed construction*: "wave that can be modulated to carry data"

   c. *Court's construction*: "signal that can be modulated to carry data"

The parties agree that "carrier signal" and "carrier" should have the same construction. (D.I. 144 at 36). Defendants argue strenuously that the proper construction for this term requires that the carrier signal be a wave and that this construction is supported by the specification itself. (*Id.* at 33). Contrary to Defendants' assertion, however, neither "wave" nor "waveform" appear anywhere in the specification. To require that the carrier be a wave, therefore, would be to import

a term that itself requires construction. Plaintiff argues that the wave Defendants refer to throughout their briefing and during oral argument is simply the time domain representation of a signal that exists only after the carrier signals are modulated and combined. (*Id.* at 21, 33, 35; D.I. 158 at 70:12-18). The specification supports Plaintiff's position, describing the carrier signals as being modulated in the frequency domain prior to being combined into the time domain transmission signal. ('158 patent at 4:12-24). While I find support for Plaintiff's opposition to using the word "wave" in the construction of this term, I agree with Defendants that some construction is needed, so I will adopt Defendants' construction modified as follows: "signal that can be modulated to carry data."

2. "determin[e/ing] a phase shift for the carrier signal"

   a. *Plaintiff's proposed construction*: "plain meaning"

   b. *Defendants' proposed construction*: "use/using an equation to compute the degrees or radians that the phase of the carrier signal can be shifted"

   c. *Court's construction*: "comput[e/ing] an amount by which the phase of the carrier signal will be shifted"

As an initial matter, the parties disagree as to whether the phase shift must be determined in units of degrees or radians. There is no support in the intrinsic record for Defendants' attempt to import these terms into the claim. Degrees and radians are merely units of measure, akin to feet or meters. I see no reason to limit this claim term to require specific units of measure for the phase shift.

Defendants next argue that this term should be construed to limit the meaning of "determine" to mean compute. Defendants cite the invention as described in the "Summary of the Invention" section of the specification as support and argue that the invention as a whole is described using the word "compute" with respect to how the phase shift is determined. (D.I. 144

at 38). I agree with Defendants. The specification, in describing the "present invention," states that "[a] phase shift is computed for each carrier signal." ('158 patent at 2:39-40). Every reference to the phase shift in the Summary of the Invention section reflects that the shift is "computed." *See id.* at 2:43, 2:58-59, 2:63-64. "When a patent thus describes the features of the 'present invention' as a whole, this description limits the scope of the invention." *Verizon Servs. Corp. v. Vonage Holdings Corp.*, 503 F.3d 1295, 1308 (Fed. Cir. 2007).

Defendants further argue that "by definition, to 'compute' is to use an equation." (D.I. 144 at 39). Plaintiff counters that the definition of compute is broader and that Defendants are "attempting to import a limitation from an example embodiment." (*Id.* at 39-40). On this point I agree with Plaintiff. Although the example embodiments do employ an equation to compute the phase shifts, the specification disclaims reliance on any particular method, stating that "additional and/or different phase shifting techniques can be used by the phase scrambler." ('158 patent at 8:14-15). Defendants also cite to the provisional application as further support for their argument; however, the provisional application also disclaims reliance on any particular method for determining the phase shifts, stating that "[t]he fundamental principle used in this invention is to use known parameters at the transmitter and the receiver to randomize the phase of the tones in a multicarrier system." (D.I. 146 at A355).

Therefore, I decline to adopt either Plaintiff's or Defendants' proposed constructions. Instead I construe the term "determin[e/ing] a phase shift for the carrier signal" to mean "comput[e/ing] an amount by which the phase of the carrier signal will be shifted."

3. "phase scrambler"

a. *Plaintiff's proposed construction*: "a component operable to adjust the phases of the carriers, by pseudo-randomly varying amounts"

b. *Defendants' proposed construction*: "component that adjusts the phases of modulated carrier signals by pseudo-randomly varying amounts"

c. *Court's construction*: "component operable to adjust the phases of the carrier signals, by pseudo-randomly varying amounts"

"scrambling the phase characteristics of the carrier signals"

a. *Plaintiff's proposed construction*: "adjusting the phase characteristics of the carrier signals by pseudo-randomly varying amounts"

b. *Defendants' proposed construction*: "adjusting the phases of the modulated carrier signals by pseudo-randomly varying amounts"

c. *Court's construction*: "adjusting the phase characteristics of the carrier signals by pseudo-randomly varying amounts"

The parties' only dispute with respect to these two claim terms is whether the carrier signals are modulated before or after phase scrambling occurs. Plaintiff argues that in every embodiment disclosed in the specification phase scrambling occurs before modulation. (D.I. 144 at 42). Defendants counter that the specification requires "adding phase shifts to modulated carrier signals." (*Id.* at 43). I find that Plaintiff's position is supported by the patent. For example, the specification describes the process that takes place in the transmitter as "adjusting the phase characteristic of each carrier signal and combining these carrier signals to produce the transmission signal." ('158 patent, 5:16-19). The specification also provides descriptions of several different phase shifting examples, and then states, "The DMT transmitter 22 then combines (step 130) the carrier signals to form the transmission signal 38." (*Id.* at 8:17-19). Defendants' attempt to parse phrases such as "method that scrambles the phase characteristics of the modulated carrier signals in a transmission signal" to require that the signals be modulated before phase scrambling is unavailing. (D.I. 144 at 48-49). This phrase, taken from the Summary of the Invention, is nothing more than a high-level description of the transmission signal as being composed of modulated carrier signals whose phases have been scrambled. Nothing in the claims or the descriptions of

example embodiments supports Defendants' argument that the phase scrambling occurs after modulation. I will adopt Plaintiff's construction.

4. "transceiver"

a. *Plaintiff's proposed construction*: "a communications device capable of transmitting and receiving data over the same physical medium wherein the transmitting and receiving functions are implemented using at least some common circuitry"

b. *Defendants' proposed construction*: "communications device with a transmitter and receiver"

c. *Court's construction*: "communications device capable of transmitting and receiving data wherein the transmitter portion and receiver portion share at least some common circuitry"

This term appears in all six of the asserted patents and the parties agree that the term should have the same construction in each claim. (D.I. 144 at 22). The parties also agree that a transceiver is a device that can both transmit and receive data. The parties dispute, however, whether the transmission and reception must occur over the same physical medium, *e.g.*, over cable or air, and whether the transmitter and receiver components of the transceiver must share common circuitry. As to the first point of dispute, there is no support in either the intrinsic or extrinsic record for the limitation that the transmission and reception of data occur over the same physical medium. Plaintiff cites only to an expert declaration to support its contention that a person of ordinary skill in the art would understand that the transmitting and receiving must occur over the same physical medium. (D.I. 144 at 25). However, nothing in the claims or specification supports this construction and Plaintiff has not pointed to any dictionary definitions or evidence other than the expert declaration to support its construction. I decline to import this limitation into the claim term.

As to the common circuitry limitation, the only information to be gleaned from the claim language itself is that the transceiver contemplated by these patents must be able to both transmit and receive data. (*See, e.g.*, '158 patent, claim 1). The specifications do not provide an explicit definition of transceiver. In the phase scrambling patents, the specification and figures indicate that the transceiver as described is a singular device housing both a transmitter portion and a receiver portion. (*Id.* at 3:31-33). These patents do not provide any specific indication that any circuitry is shared between the two. In the low power mode patents, however, the specification and figure do indicate the presence of shared components. For example, the clock, controller, and frame counter are shared by the transmitter and receiver portions of the transceiver. ('404 patent at Fig. 1).

The parties provide five different dictionary definitions for transceiver, three of which include a limitation that the transmitter and receiver share common circuitry. (D.I. 146 at A423, A433, A444, A891, A938-39). Evaluating the intrinsic evidence in light of these dictionary definitions suggests that the transmitter and receiver portions do share common circuitry or components. Therefore, I will construe transceiver to mean "a communications device capable of transmitting and receiving data wherein the transmitter portion and receiver portion share at least some common circuitry."

5. "multicarrier"

a. *Plaintiff's proposed construction*: "having multiple carrier signals that are combined as a group by simultaneous modulation to produce a transmission signal"

b. *Defendants' proposed construction*: "having multiple carrier signals that are combined to produce a transmission signal"

c. *Court's construction*: "having multiple carrier signals that are combined to produce a transmission signal"

The parties' only disagreement is whether this term should be construed to specify a particular method by which the carrier signals are combined. Plaintiff's opposition to Defendants' broader construction appears to stem from its disagreement with Defendants' proposed construction of "carrier." (D.I. 158 at 30:20-31:7). Since I have rejected Defendants' proposed limitations on "carrier," this concern is unwarranted. As discussed above, I have concluded that the patents disclose combination and modulation of carrier signals in the frequency domain, that is, before a time domain signal, or wave, exists. Turning to Plaintiff's proposed limitation, I find that the claim language itself does not impose any limitation on how the carrier signals are to be combined. Nor does the specification provide such limitations. Therefore, I will adopt Defendants' proposed construction.

6. "bit scrambler"

a. *Plaintiff's proposed construction*: "a component that pseudo-randomly changes the value of a bit"

b. *Defendants' proposed construction*: "component that pseudo-randomly inverts the bits in a byte of data one bit after another"

c. *Court's construction*: "component that pseudo-randomly changes the value of a bit"

The parties disagree on two points in their proposed constructions of this term: first, whether the bit scrambler operates on a byte of data; and second, whether the bits are scrambled in sequence, one after another. The parties' disagreement appears to center around whether a person of ordinary skill in the art would find that a bit scrambler is different from a byte scrambler. I do not think it is necessary to resolve this disagreement as the patent itself provides sufficient guidance as to the meaning of "bit scrambler."

The word "byte" does not appear in either the claims or specification of the '243 patent. The patent refers to "scrambling, using the bit scrambler, a plurality of input bits." ('243 patent,

claim 1). A plurality of input bits simply means more than one input bit. A byte of data is commonly understood to consist of eight bits of data. *See, e.g.*, OXFORD ENGLISH DICTIONARY (2d ed. 1989), *available at* http://www.oed.com/oed2/00030648 (defining byte as "[a] group of eight consecutive bits operated on as a unit in a computer"). There is no basis in the claim itself or in the specification for requiring that the "plurality of input bits" consist of a byte, or eight bits, of data. Nor is there any indication in the patent that the data must be presented to the scrambler a byte at a time. Rather, as Defendants themselves point out, the data is presented a bit at a time. Defendants cite the ADSL standards as extrinsic evidence of what a person of ordinary skill would understand a "bit scrambler" to be. (D.I. 144 at 52-53). The device described in the standards, however, is simply called a "scrambler," not a "bit scrambler." (D.I. 145 at A503). Furthermore, the standards show that data is input to this scrambler a byte at a time, not as a serial bit stream. (*Id.*). This is inconsistent with the bit scrambler described in the specification.

As to Defendants' argument that the scrambling must be performed sequentially, the claim language does not support such a limitation. The claim itself is indifferent to whether the scrambling is sequential, stating that the bit scrambler scrambles "a plurality of input bits to generate a plurality of output bits." (*Id.*). The specification states that the bit scrambler "receives the input serial bit stream" and, after scrambling, passes the bits to the QAM encoder. (*Id.* at 5:6-9). The QAM encoder is described as "receiving an input serial data bit stream." (*Id.* at 3:63-64). This seems to indicate that the input and output of the bit scrambler are both serial. This does not mean, however, that the scrambling itself necessarily takes place sequentially. Therefore, the intrinsic evidence does not support Defendants' proposed limitations and I will adopt Plaintiff's construction.

B. The Low Power Mode Patents

The '404 patent is directed to a multicarrier transmission system with low power sleep mode and rapid-on capability. Claim 6 is representative and reads as follows:

> 1. An *apparatus comprising a transceiver operable to*:
> receive, in a full power mode, a plurality of superframes, wherein the superframe comprises a plurality of *data* frames followed by a *synchronization frame*;
> receive, in the full power mode, a *synchronization signal*;
> transmit a message to enter into a *low power mode*;
> *store, in the low power mode, at least one parameter* associated with the full power mode operation *wherein the at least one parameter comprises at least one of a fine gain parameter and a bit allocation parameter*;
> receive, in the *low power mode*, a *synchronization signal*; and
> exit from the *low power* [sic] and restore the full power mode by using the at least one parameter and without needing to reinitialize the transceiver.

('404 patent, claim 6) (disputed terms italicized).

The '268 patent is also directed to a multicarrier transmission system with low power sleep mode and rapid-on capability. Claim 4 is representative and reads as follows:

> 4. A method, in a multicarrier transceiver, comprising:
> transmitting or receiving a message to enter a *low power mode*;
> entering the *low power mode*, wherein a transmitter portion of the transceiver does not transmit *data* during the *low power mode* and a receiver portion of the transceiver receives *data* during the *low power mode*; and
> *storing, during the low power mode, at least one parameter* associated with a full power mode.

('268 patent, claim 4) (disputed terms italicized).

1. "low power mode"

   a. *Plaintiff's proposed construction*: "a state of operation in which power is consumed, but the amount of power consumed is less than when operating in a state with full data transmission capabilities"

   b. *Defendants' proposed construction*: "state of operation in which available power is reduced"

c. *Court's construction*: "state of operation in which power is consumed, but the amount of power consumed is less than when operating in a state with full data transmission capabilities"

The primary dispute between the parties with respect to this term appears to center on whether low power mode requires that less power be supplied to the circuitry or whether less power is consumed by the device. The parties also disagree about whether the claimed "low power mode" includes both the "sleep mode" and "idle state/mode" described in the specification.

Neither sleep mode nor idle state/mode are mentioned in any of the claims. Defendants expended significant effort both in briefing and at oral argument to argue that "idle state" is not a low power mode. I disagree. The specification states in a number of different places that the invention could be incorporated into a computer and that it would be desirable in that situation that it could "enter a 'sleep' mode in which it consumes reduced power." ('404 patent at 6:2-3). The specification describes this as an "'idle' state . . . similar in many ways to the sleep mode state." (*Id.* at 6:19:24). Defendants argue that it is significant that the specification sometimes calls this a "state" instead of a "mode." (D.I. 158 at 21:10-22:3). I do not think so. Elsewhere in the specification, the same idle state is referred to as an "idle mode." ('404 patent at 8:63). It seems to me that sleep mode and idle state/mode are both low power modes implemented in different contexts.

The dispute over whether low power mode is achieved through lower power consumption or lower power supply is readily resolved by looking to the claim language. Low power mode appears in independent claims 1, 6, 11, and 16 of the '404 patent. Although claim 1 of the '404 patent is not asserted, "we look to the words of the claims themselves, both asserted and nonasserted, to define the scope of the patented invention." *Vitronics Corp. v. Conceptronic, Inc.*, 90 F.3d 1576, 1582 (Fed. Cir. 1996). Claims 1 and 11 read, in part, "[enter/entering] into the low

power mode by reducing a power consumption of at least one portion of a transmitter." ('404 patent, claims 1 & 11). Claims 6 and 16 do not include this phrase describing how low power mode is achieved. "Unless the patent otherwise provides, a claim term cannot be given a different meaning in the various claims of the same patent." *Georgia-Pac. Corp. v. U.S. Gypsum Co.*, 195 F.3d 1322, 1331 (Fed. Cir. 1999). Read in the context of the specification, I find no reason why the term should be given different meaning in claims 6 and 16 than it has in claims 1 and 11, which indicate that low power mode is achieved through lower consumption of power.

Finally, the parties dispute whether low power mode includes, as Defendants argue, a state in which the device is completely off. (D.I. 144 at 61). Defendants' argument on this point is inconsistent with the claims and specification. While in low power mode, the transceiver must be able to either transmit or receive a synchronization signal. ('404 patent, claims 1 and 6). The argument that some power is consumed by the transceiver even in low power mode is supported by the specification. (*Id.* at 7:44-56). For these reasons, I will adopt Plaintiff's construction.

2. "stor[e/ing], in [a/the] low power mode, at least one parameter"

a. *Plaintiff's proposed construction*: "maintaining in memory at least one parameter associated with a mode of operation with full data transmission capabilities, while in a low power mode"

b. *Defendants' proposed construction*: "maintain[ing] in memory throughout a/the low power mode, at least one parameter"

c. *Court's construction*: "maintain[ing] in memory, while in low power mode, at least one parameter"

The parties first dispute whether the construction should include the limitation that the parameter must be associated with full power mode. Defendants argue that this limitation already appears in the claim language and including this in the claim construction would be superfluous. (D.I. 144 at 66). Plaintiff did not reply to this argument. I agree with Defendants. The claim

language includes this limitation already when it calls for storing "at least one parameter associated with the full power mode operation." ('404 patent, claim 6). It would be redundant to include this in the court's construction of this term.

The parties also disagree about whether the parameter must be maintained throughout the duration of the low power mode. Plaintiff argues that there is no support in the claim language for requiring a particular duration for how long the parameter is stored. (D.I. 144 at 65). Defendants counter that it is a "fundamental requirement" of the invention that the parameter be stored for the entire duration of the low power mode. (*Id.*). Reading the claim as a whole, I find it is unnecessary to include this requirement in the construction of this term. The claim specifies that the device will "exit from the low power mode and restore the full power mode by using the at least one parameter." ('404 patent, claim 6). Therefore, the rest of the claim itself implies that the parameter is stored at least until the device exits from low power mode. This is captured by the court's construction of "maintain[ing] in memory, while in low power mode, at least one parameter."

3. "wherein the at least one parameter comprises at least one of a fine gain parameter and a bit allocation parameter"

a. *Plaintiff's proposed construction*: "wherein the at least one parameter includes a fine gain parameter and/or a bit allocation parameter"

b. *Defendants' proposed construction*: "wherein the at least one parameter includes both a fine gain parameter and a bit allocation parameter"

c. *Court's construction*: "wherein the at least one parameter includes a fine gain parameter and/or a bit allocation parameter"

Plaintiff argues that its construction follows the plain language of the claim and notes that the parameters listed in the claim are not categories but rather two parameters from a list of parameters that may be stored. (D.I. 144 at 90-92). Defendants argue that the phrase "at least one of" modifies both terms, requiring that both a fine gain and a bit allocation parameter must be

stored, citing Federal Circuit case law in support of their position. (*Id.* at 91). Defendants are correct that the Federal Circuit has previously construed this same phrase to require one of each of the terms in the list as a matter of grammatical construction. *SuperGuide Corp. v. DirecTV Enterprises, Inc.*, 358 F.3d 870, 886 (Fed. Cir. 2004). As a number of district courts have recognized, however, "*SuperGuide* did not erect a universal rule of construction for all uses of 'at least one of' in all patents." *Fujifilm Corp. v. Motorola Mobility LLC*, 2015 WL 1265009, at *8 (N.D. Cal. Mar. 19, 2015).

I find that this phrase is readily construed by looking at the full context of the claim itself, without having to resort to grammatical arguments. The relevant portion of the claim reads "storing, in the low power mode, *at least one parameter* associated with the full power mode operation wherein the at least one parameter comprises at least one of a fine gain parameter and a bit allocation parameter." ('404 patent, claim 11 (emphasis added)). The phrase "at least one parameter" indicates that the patent contemplates a situation where only one parameter would be stored. Defendant's construction would require a minimum of two parameters to be stored and is, therefore, inconsistent with the plain language of the claim. For this reason, I will adopt Plaintiff's construction.

4. "fine gain parameter"

   a. *Plaintiff's proposed construction*: "a parameter used to determine power level on a per subcarrier basis"

   b. *Defendants' proposed construction*: "Indefinite"

   c. *Court's construction*: "parameter used to determine power level on a per subcarrier basis"

Defendants only argument with respect to this term is that "fine" is a word of degree and, therefore, this term is necessarily indefinite. (D.I. 144 at 68). I disagree. The claim language does

not instruct that anything be measured or adjusted, as in, "make a fine adjustment to the gain," for example. Rather, the claim instructs that a specific parameter, named the "fine gain parameter," is to be stored. Although the claim language itself does not provide specific guidance as to the meaning of this term, the specification supports Plaintiff's construction, particularly when considered in the context of the extrinsic evidence Plaintiff presents to show that a person of ordinary skill in the art would understand that "fine gain" refers to the gain on a subchannel. For example, the specification discusses the requirements of initialization, and in doing so distinguishes between "setting the channel gains" and "adjusting the fine gains on the subchannels." ('404 patent at 3:12-14). This distinction is substantially supported by the ITU-T G.992.1 Standards Plaintiff referenced in its briefing and presented at oral argument as evidence of what a person of ordinary skill in the art would understand "fine gain" to mean.[2] (D.I. 144 at 70; D.I. 190 at 121:17-122:5). Therefore, I will adopt Plaintiff's construction.

5. "bit allocation parameter"

a. *Plaintiff's proposed construction*: "parameter used to determine a number of bits to be carried by a subcarrier on a per subcarrier basis"

b. *Defendants' proposed construction*: "parameter specifying the number of bits to be carried by a subchannel"

c. *Court's construction*: "parameter used to determine a number of bits to be carried by a subcarrier on a per subcarrier basis"

The parties have two disputes in construing this term. First, they disagree on whether the parameter is used to determine the number of bits or whether it specifies the number of bits. Second, they dispute whether the parameter provides the number of bits carried by a single subcarrier or whether it provides the number of bits on a per subcarrier basis, i.e. whether the Bit

[2] The relevant time period for this understanding is January 26, 1998, the priority date of both the '404 and '268 patents.

Allocation Table referenced in the specification is itself a bit allocation parameter. As to the first dispute, limiting the term to mean "specifying" would encompass how the number of bits is determined when a Bit Allocation Table is used, as described in the exemplary embodiment. The word "determine" also encompasses the use of a table to perform this task. Defendants argue that using "determine" unduly broadens the definition. I disagree. Only if I were to limit the claim to require that the only form of a bit allocation parameter be a Bit Allocation Table would Defendants' argument carry the day. The specification describes a method for constructing the Bit Allocation Table. But it is a parameter and not the Table itself that is claimed. It is not difficult to imagine other methods of determining the number of bits to be carried that do not involve a Bit Allocation Table being the parameter that is stored. Thus, I do not limit the construction to the exemplary embodiment.

The second dispute is readily resolved by turning to the specification. The patent lists some of the requisite parameters for waking from sleep mode and "Bit Allocation Tables" is included in that list. ('404 patent at 8:6-12). It seems to me that a full Bit Allocation Table is one example of the bit allocation parameter referenced in the claims. Therefore, Defendants' argument that a bit allocation parameter is nothing more than a single entry in a Bit Allocation Table must fail. Plaintiff's position that the number of bits must be specified for each subcarrier, not just a single subcarrier, is supported by the specification and comports with the purpose of the invention, i.e., allowing a transceiver to wake from sleep mode without reinitializing. Furthermore, the claim does not limit the form of the parameter to only a Bit Allocation Table. Therefore, I will adopt Plaintiff's construction.

6. "synchronization frame"

a. *Plaintiff's proposed construction*: "a frame that indicates a superframe boundary"

b. *Defendants' proposed construction*: "frame that carries no user or overhead bit-level data and is inserted to establish superframe boundaries"

c. *Court's construction*: "frame that indicates a superframe boundary"

The parties agree that synchronization frames indicate or establish superframe boundaries. The parties disagree, however, as to whether the synchronization frame must be limited to that defined in the ITU Document G922.2. Defendants insist that it must be so limited, pointing to the specification, which references this ITU Document. ('404 patent at 5:5-12). There are two problems with Defendants' argument, however. First, the reference to the ITU document is made after the reference to data frames and is also given specifically as an example ("data frames (*e.g.*, sixty-eight frames for ADSL as specified in ITU Document G.992.2)"). No reference is made to the ITU document after the synchronization frame is mentioned. Second, this is a simply an exemplary embodiment and I find no evidence to support limiting the claim to one exemplary embodiment. Therefore, I will adopt Plaintiff's construction.

7. "synchronization signal"

a. *Plaintiff's proposed construction*: "an indication used to establish or maintain a timing relationship between transceivers"

b. *Defendants' proposed construction*: "reference wave used to establish or maintain a timing relationship between transceivers"

c. *Court's construction*: "signal used to establish or maintain a timing relationship between transceivers"

The only dispute between the parties with respect to this term is whether the signal is "an indication" or a "reference wave." Defendant argues strenuously that the signal must be a wave, arguing that all of the examples of synchronization signals given in the specification are "reference waves." (D.I. 144 at 81). Defendant does not explain, however, what exactly a reference wave is in this context. The phrase "reference wave" does not appear anywhere in the patent and

Defendant has offered no definition. I will not construe this claim term to include a phrase that adds ambiguity and uncertainty to the meaning of the term. Plaintiff's proposal of "indication," however, is little better as the word "indication" could easily be deemed to include things that are not "signals." It seems to me that "signal" is a well-understood term that has a plain meaning to those skilled in the art. I see no need to substitute a different word that would introduce ambiguity into the meaning of the term. Therefore, I will adopt Plaintiff's proposed construction, modified as follows: "signal used to establish or maintain a timing relationship between transceivers."

8. "apparatus comprising a transceiver operable to"

   a. *Plaintiff's proposed construction*: "See above for the construction of 'transceiver'; otherwise plain meaning"

   b. *Defendants' proposed construction*: "The preamble is limiting[3] and this is a means-plus-function limitation. The "transceiver" is the CPE transceiver depicted in Figure 2"

   c. *Court's construction*: "plain meaning with 'transceiver' as previously construed"

Defendants argue that this element from the preamble of several claims is limiting as a means-plus-function claim element governed by 35 U.S.C. § 112 ¶ 6 because the word transceiver does not impart definite structure. (D.I. 144 at 85). Plaintiff responds that transceiver has a well-understood structural meaning in the art. (*Id.* at 86). When the word "means" does not appear in the claim element, there is a presumption that the element is not means-plus-function. *Williamson v. Citrix Online, LLC*, 792 F.3d 1339, 1349 (Fed. Cir. 2015). "[T]he presumption can be overcome and § 112, para. 6 will apply if the challenger demonstrates that the claim term fails to 'recite sufficiently definite structure' or else recites 'function without reciting sufficient structure for performing that function.'" *Id.*

[3] Defendants argue only that the preamble provides a functional limitation. Therefore, I decline to address whether the preamble is otherwise limiting.

I conclude that § 112 ¶ 6 does not apply to this claim element. The word "means" does not appear in the claim element, so I begin with the presumption that § 112 ¶ 6 does not apply. Defendants have not overcome this presumption. Although "apparatus" is a non-structural term, the word "transceiver" imparts sufficient structure to the claim element. Transceiver is not a generic term like module or device. *Id.* at 1350. Rather, transceiver is the name of a device well known in the field of communications and, furthermore, the claimed transceiver is sufficiently described in the specification. (*See* '404 patent at 4:14-5:36). I will adopt Plaintiff's construction.

9. "data"

a. *Plaintiff's proposed construction*: "non-control information"

b. *Defendants' proposed construction*: "digital information"

c. *Court's construction*: "content"

Plaintiff initially argued that this term should be construed to have its plain meaning. (D.I. 144 at 87-88). Plaintiff proposed "non-control information" in response to Defendants' initial proposed construction, "information." (*Id.* at 89). At oral argument, Defendants proposed to narrow their construction to "digital information." (D.I. 190 at 144:21). I am not persuaded that any of these constructions provide any clarity as to the meaning of the term "data." At oral argument, I proposed construing the term to mean "content." (*Id.* at 151:21). Plaintiff agreed to this proposed construction. (*Id.* at 155:9-156:1).

Defendants, however, argue that construing data to mean "content" would impermissibly narrow the meaning of "data" in some of the claims because "user data" is used in other claims. (*Id.* at 156:4-12). According to Defendants, user data is content. This position is contradicted by the patent specification, however. The specification provides that during sleep mode, "user data provided by the CO transceiver will be benign idle data such as ATM IdleCells or HDLC Flag

octets." ('268 patent at 7:34-36). Although this information is defined to be user data by the patent itself, it is not content. Therefore, I will construe data to mean content.

C. The Diagnostic Mode Patents

The '430 patent is directed to multicarrier modulation messaging for frequency domain received idle channel noise information. Claim 1 is representative and reads as follows:

> 1. A transceiver capable of *transmitting test information over a communication channel* using multicarrier modulation comprising:
>
> a transmitter portion capable of transmitting a message, wherein the message comprises one or more data variables that represent the *test information*, wherein bits in the message are modulated onto DMT symbols using Quadrature Amplitude Modulation (QAM) with more than 1 bit per subchannel and wherein at least one data variable of the one or more data variables comprises an *array representing frequency domain received idle channel noise information*.

('430 patent, claim 1) (disputed terms italicized).

The '412 patent is directed to multicarrier modulation messaging for power level per subchannel information. Claim 1 is representative and reads as follows:

> 1. A transceiver capable of *transmitting test information over a communication channel* using multicarrier modulation comprising:
>
> a transmitter portion capable of transmitting a message, wherein the message comprises one or more data variables that represent the *test information*, wherein bits in the message are modulated onto DMT symbols using Quadrature Amplitude Modulation (QAM) with more than 1 bit per subchannel and wherein at least one data variable of the one or more data variables comprises an *array representing power level per subchannel information*.

('412 patent, claim 1) (disputed terms italicized).

1. "[transmitting/receiving] test information over a communication channel"

   a. *Plaintiff's proposed construction*: "plain meaning"

   b. *Defendants' proposed construction*: "transmitting/receiving test information to/from a central office modem"

   c. *Court's construction*: "plain meaning"

Defendants seek to import a limitation into this claim requiring that the test information be transmitted either to or from a central office modem. This limitation is unsupported by either the claims or the specification. The specification does indicate that the receiving transceiver is "typically located" at the central office, but typically does not mean always. ('412 patent at 1:53). Defendants argue that the patent is directed to the solution of a particular problem: diagnosing problems without the need to dispatch a technician to the customer's home. (D.I. 144 at 97). This may be a problem identified in the specification that is solved by this patent, but the solution to the problem is not so limited. I find no basis for importing this limitation into the claim. I agree with Plaintiff that this term should be given its plain meaning. Defendants are prohibited from arguing that the term is limited to communications over a channel that includes the central office modem.

2. "test information"

a. *Plaintiff's proposed construction*: "information relating to a measured characteristic of a communication channel"

b. *Defendants' proposed construction*: "information relating to a disturbance in the communication channel"

c. *Court's construction*: "information relating to a characteristic of a communication channel or the communications equipment operating on that channel"

The parties dispute whether the test information must be measured and whether the information must relate to a disturbance in the communications channel. I find that neither of these limitations is supported by the intrinsic evidence.

Defendants contend that the description of the invention as a whole in the specification is limiting and that test information must therefore be limited to information "relate[d] to the diagnosis and resolution of communications problems caused by a disturbance on a communications channel." (D.I. 144 at 102). Defendants' argument is unavailing. The

specification states, "The systems and methods of this invention are directed toward reliably exchanging diagnostic and test information between transceivers over a digital subscriber line in the presence of voice communications and/or other disturbances." ('430 patent at 1:44-47). Nothing in this description provides any limitation on the definition of test information. The reference to disturbances means only that the invention provides a method for the exchange of test information when there is a disturbance on the line. The specification later provides an extensive, but not exhaustive, list of what test information might include. (*Id.* at 2:24-43). Many of the items in this list are unrelated to disturbances. It would be inappropriate to limit the definition of test information when nothing in the specification indicates such a limitation.

With respect to whether the information must be measured, Plaintiff argues that a person of ordinary skill in the art would recognize that the test information as claimed must be measured. (D.I. 144 at 105). Defendants counter that the specification includes a list of categories of information that may be included as the test information and that a number of the items on the list, such as Chip Type, do not require measurement to determine. (*Id.* at 104). I agree with Defendants. Although some types of test information, as defined in the specification, must be measured, other types are simply characteristics of the communications system.

Defendants further challenge Plaintiff's construction as improperly limiting the test information to characteristics of a communications channel. (*Id.*) Defendants point out that information such as Chip Type and Vendor ID are characteristics of the modems, not of the communications channel itself. (*Id.*). I agree with Defendants. The test information defined in the specification appears to more broadly encompass information related not only to the communications channel itself, but also to the equipment used at one end of the channel. Therefore, I will adopt the following construction for test information: "information relating to a

characteristic of a communication channel or the communications equipment operating on that channel."

3. "array representing frequency domain received idle channel noise information"

   a. *Plaintiff's proposed construction*: "ordered set of values representative of noise in the frequency domain measured on respective subchannels while no input signals are being transmitted on the subchannels"

   b. *Defendants' proposed construction*: "ordered set of values representative of noise in the frequency domain that was received by a transceiver on a channel in the absence of a transmission signal"

   c. *Court's construction*: "ordered set of values representative of noise in the frequency domain that was received by a transceiver on respective subchannels in the absence of a transmission signal"

The parties have three disputes with respect to this term: whether the values must be measured; whether the values represent noise on a subchannel basis; and whether the idle channel noise corresponds to "no input signals" being transmitted or simply "the absence of a transmission signal." The first and third disputes are readily resolved. There is no indication, either in the claims or in the specification, as to how these values are obtained. Certainly the values may be measured, but I cannot find support in the intrinsic evidence to limit the construction to measured values only. Furthermore, Plaintiff's own extrinsic evidence, and the only evidence presented with respect to the meaning of "idle channel noise," indicates that Defendants propose the better construction. *See* NEWTON'S TELECOM DICTIONARY 410 (15th ed. 1999) (defining idle channel noise as "[n]oise which exists in a communications channel when no signals are present"). There is no support for limiting idle channel noise to noise present in the absence of "input signals." Therefore, as to these two disputes, I adopt Defendants' proposed construction.

As to the dispute over whether the values are measured on respective subchannels, I find Defendants' arguments unavailing. Defendants are correct to point out that the applicants used the

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phrase "per subchannel" explicitly in the '412 patent. ('412 patent, claim 1). However, the "array" terms of the two patents are differently worded. Thus, the absence of this phrase in the claims of the '430 patent does not necessarily render the phrase superfluous in the '412 patent. Furthermore, the fact that what is claimed is an "array" implies that more than one value is included. Therefore, I decline to adopt either party's proposed construction and instead will construe this term to mean "ordered set of values representative of noise in the frequency domain that was received by a transceiver on respective subchannels in the absence of a transmission signal."

4. "array representing power level per subchannel information"

a. *Plaintiff's proposed construction*: "ordered set of values representative of power levels measured on respective subchannels"

b. *Defendants' proposed construction*: "ordered set of values representative of power levels of respective subchannels"

c. *Court's construction*: "ordered set of values representative of power levels of respective subchannels"

The parties' only dispute with respect to this term is whether the values must be measured. Plaintiff argues that without specifying that the values are measured, the term could be understood to mean that the values represent power level settings. (D.I. 144 at 115). Plaintiff further argues that the very definition of test information requires that the values be measured. (*Id.* at 116). I have already rejected the argument that all test information must be measured, however. Plaintiff cites to dependent claims specifying that the power levels are "based on a Reverb signal" and, therefore, must be measured. (*Id.*). Plaintiff further points to the specification, which provides that the power levels are "detected during the ADSL Reverb signal." (*Id.*). Defendants counter that detecting is not the same as measuring and that nothing in the claims or specification require that "the *only* way to obtain power level information is to measure it." (*Id.* at 117). Defendants

further argue that there is a presumption that a limitation present in a dependent claim is not present in the independent claim. (*Id.*).

As an initial matter, I reject Defendants' argument that detect and measure have different meanings in this context. I do, however, agree with Defendants argument that the limitation in the dependent claim should not be imported into the independent claim. Plaintiff's citations to the specification describe a preferred embodiment which, it seems to me, directly corresponds with the dependent claims. While I do not see any reason these power levels could not be measured, or that they must be obtained in any particular way, I also do not see any support for requiring that they be measured. Therefore, I will adopt Defendants' proposed construction.

5. "Reverb signal"

a. *Plaintiff's proposed construction*: "a signal generated by modulating carriers in a multicarrier system with a known pseudo-random sequence to generate a wideband modulated signal"

b. *Defendants' proposed construction*: "any 'REVERB' signal defined in the ITU or ANSI ADSL standards in existence as of January 8, 2001"

c. *Court's construction*: "signal generated by modulating carriers in a multicarrier system with a known pseudo-random sequence to generate a wideband modulated signal"

The primary dispute between the parties with respect to this construction is whether, as Defendants argue, the Reverb signal is limited to that defined in the referenced standards. Defendants find support for this limitation both in the fact that the term is capitalized, which Defendants take to indicate a reference to the REVERB1 signal from the standards, as well as from the reference to the standards in the specification. (D.I. 144 at 112-13). I find Defendants' argument unconvincing. Although the term "Reverb" is capitalized in the claims, it is not spelled out in all capital letters, nor does it include the number "1" at the end. Everywhere the specific standard is mentioned in the specification, it is given as "REVERB1." ('412 patent at 3:57-4:3).

If the applicant had meant to claim the specific REVERB1 signal from the relevant standards, it seems likely he would have named that specific signal in the claim. The specification refers to the REVERB1 signal from the standards when describing an exemplary embodiment and there is no evidence in the specification of any disclaimer of other ways of generating a Reverb signal.

Plaintiff's proposed construction, on the other hand, is drawn directly from the specification. (*Id.* at 3:62-64). The applicant chose to define how the Reverb signal was to be generated. Having found no compelling reason to impose additional limitations on the meaning of this term, I will adopt Plaintiff's construction.

IV. CONCLUSION

Within five days the parties shall submit a proposed order consistent with this Memorandum Opinion suitable for submission to the jury.

IN THE UNITED STATES DISTRICT COURT
FOR THE DISTRICT OF DELAWARE

TQ DELTA, LLC, Plaintiff, v. COMCAST CABLE COMMUNICATIONS LLC, Defendant.	C.A. No. 15-cv-611-RGA
TQ DELTA, LLC, Plaintiff, v. COXCOM LLC and COX COMMUNICATIONS INC., Defendants.	C.A. No. 15-cv-612-RGA
TQ DELTA, LLC, Plaintiff, v. DIRECTV, LLC, Defendant.	C.A. No. 15-cv-613-RGA
TQ DELTA, LLC, Plaintiff, v. DISH NETWORK CORPORATION, DISH NETWORK L.L.C., DISH DBS CORPORATION, ECHOSTAR CORPORATION, AND ECHOSTAR TECHNOLOGIES L.L.C., Defendants.	C.A. No. 15-cv-614-RGA

TQ DELTA, LLC, Plaintiff, v. TIME WARNER CABLE INC. and TIME WARNER CABLE ENTERPRISES LLC, Defendants.	C.A. No. 15-cv-615-RGA
TQ DELTA, LLC, Plaintiff, v. VERIZON SERVICES CORP., Defendant.	C.A. No. 15-cv-616-RGA

CLAIM CONSTRUCTION ORDER

The Court has determined that the terms below shall be given the following meaning for the claims of each identified patent:

U.S. Patent Nos. 8,718,158 and 9,014,243:

1. "**carrier signal**" and "**carrier**" – "signal that can be modulated to carry data"
2. "**determin[e/ing] a phase shift for the carrier signal**" – "comput[e/ing] an amount by which the phase of the carrier signal will be shifted"
3. "**phase scrambler**" – "component operable to adjust the phases of the carrier signals, by pseudo-randomly varying amounts"
4. "**scrambling the phase characteristics of the carrier signals**" – "adjusting the phase characteristics of the carrier signals by pseudo-randomly varying amounts"

5. "**transceiver**" – "communications device capable of transmitting and receiving data wherein the transmitter portion and receiver portion share at least some common circuitry"
6. "**multicarrier**" – "having multiple carrier signals that are combined to produce a transmission signal"
7. "**bit scrambler**" – "component that pseudo-randomly changes the value of a bit"

U.S. Patent Nos. 8,611,404 and 9,094,268:

8. "**transceiver**" – "communications device capable of transmitting and receiving data wherein the transmitter portion and receiver portion share at least some common circuitry"
9. "**multicarrier**" – "having multiple carrier signals that are combined to produce a transmission signal"
10. "**low power mode**" – "state of operation in which power is consumed, but the amount of power consumed is less than when operating in a state with full data transmission capabilities"
11. "**stor[e/ing], in [a/the] low power mode, at least one parameter**" – "maintain[ing] in memory, while in low power mode, at least one parameter"
12. "**wherein the at least one parameter comprises at least one of a fine gain parameter and a bit allocation parameter**" – "wherein the at least one parameter includes a fine gain parameter and/or a bit allocation parameter"
13. "**fine gain parameter**" – "parameter used to determine power level on a per subcarrier basis"

14. "**bit allocation parameter**" – "parameter used to determine a number of bits to be carried by a subcarrier on a per subcarrier basis"
15. "**synchronization frame**" – "frame that indicates a superframe boundary"
16. "**synchronization signal**" – ""signal used to establish or maintain a timing relationship between transceivers"
17. "**apparatus comprising a transceiver operable to**" – "plain meaning with 'transceiver' as previously construed"
18. "**data**" – "content"

U.S. Patent Nos. 7,835,430 and 8,238,412:

1. "**transceiver**" – "communications device capable of transmitting and receiving data wherein the transmitter portion and receiver portion share at least some common circuitry"
2. "**multicarrier**" – "having multiple carrier signals that are combined to produce a transmission signal"
3. "**[transmitting/receiving] test information over a communication channel**" – "plain meaning"
4. "**test information**" – "information relating to a characteristic of a communication channel or the communications equipment operating on that channel"
5. "**array representing frequency domain received idle channel noise information**" – "ordered set of values representative of noise in the frequency domain that was received by a transceiver on respective subchannels in the absence of a transmission signal"
6. "**array representing power level per subchannel information**" – "ordered set of values representative of power levels of respective subchannels"

7. "**Reverb signal**" – "signal generated by modulating carriers in a multi carrier system with a known pseudo-random sequence to generate a wideband modulated signal"

IT IS SO ORDERED this 6 day of December, 2016.

Richard G. Andrews
The Honorable Richard G. Andrews

ANSI/IEEE Std 100-1988

IEEE Standard Dictionary of Electrical and Electronics Terms

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How to Use This Dictionary

The terms defined in this dictionary are listed in alphabetical order. Terms made up of more than two words appear in the order most familiar to the people who use them. In some cases cross-references are given.

Some terms take on different meanings in different fields. When this happens the different definitions are numbered, identified as to area of origin, coded, and listed under the main entry.

If a reader wants to know the source of a definition he need only look up the code number following the definition in the SOURCES section that appears at the back of the book between pages 1112 and 1129.

transadmittance, forward (electron tubes). The complex quotient of (1) the fundamental component of the short-circuit current induced in the second of any two gaps and (2) the fundamental component of the voltage across the first. 125

trans-μ-factor (multibeam electron tubes). The ratio of (1) the magnitude of an infinitesimal change in the voltage at the control grid of any one beam to (2) the magnitude of an infinitesimal change in the voltage at the control grid of a second beam. The current in the second beam and the voltage of all other electrodes are maintained constant. 125

transceiver (1)(data transmission). The combination of radio transmitting and receiving equipment in a common housing, usually for portable or mobile use, and employing common circuit components for both transmitting and receiving. 59

(2)(navigation aid terms). A combination transmitter and receiver in a single housing, with some components being used by both parts. *See:* **transponder.** 526

transconductance. The real part of the transadmittance. *Note:* Transconductance is, as most commonly used, the interelectrode transconductance between the control grid and the plate. At low frequencies, transconductance is the slope of the control-grid-to-plate transfer characteristic. *See:* **electron-tube admittances; interelectrode transconductance.** 125

transconductance meter (mutual-conductance meter). An instrument for indicating the transconductance of a grid-controlled electron tube. *See:* **instrument.** 328

transcribe (electronic computation). To convert data recorded in a given medium to the medium used by a digital computing machine or vice versa. 235

transcriber (electronic computation). Equipment associated with a computing machine for the purpose of transferring input (or output) data from a record of information in a given language to the medium and the language used by a digital computing machine (or from a computing machine to a record of information). 210

transducer (1)(electrical heating applications to melting furnaces and forehearths in the glass industry). A device that is actuated by power from one system and supplies power in any other form to a second system. 520

(2) (communication and power transmission). A device by means of which energy can flow from one or more transmission systems or media to one or more other transmission systems or media. *Note:* The energy transmitted by these systems or media may be of any form (for example, it may be electric, mechanical, or acoustical), and it may be of the same form or different forms in the various input and output systems or media. 111,255,54

(3) (metering). A device to receive energy from one system and supply energy, of either the same or of a different kind, to another system, in such a manner that the desired characteristics of the energy input appear at the output. 212

(4) (thyristor). A device which under the influence of a change in energy level of one form or in one system, produces a specified change in energy level of another form or in another system. 445

transducer, active. A transducer whose output waves are dependent upon sources of power, apart from that supplied by any of the actuating waves, which power is controlled by one or more of the waves. *Note:* The definition of active transducer is a restriction of the more general **active network:** that is, one in which there is an impressed driving force. *See:* **transducer.** 210

transducer gain (1) (general). The ratio of the power that the transducer delivers to the specified load under specified operating conditions to the available power of the specified source. *Notes:* (A) If the input and.or output power consist of more than one component, such as multifrequency signals or noise, then the particular components used and their weighting must be specified. (B) This gain is usually expressed in decibels. *See:* **transducer.** 210

(2) (two-port linear transducer). At a specified frequency, the ratio of (A) the actual signal power transferred from the output port of the transducer to its load, to (B) the available signal power from the source driving the transducer. 125

transducer, ideal (for connecting a specified source to a specified load). A hypothetical passive transducer that transfers the maximum available power from the source to the load. *Note:* In linear transducers having only one input and one output, and for which the impedance concept applies, this is equivalent to a transducer that (1) dissipates no energy and (2) when connected to the specified source and load presents to each its conjugate impedance. *See:* **transducer.** 210

transducer, line. *See:* **line transducer.**

transducer loss. The ratio of the available power of the specified source to the power that the transducer delivers to the specified load under specified operating conditions. *Notes:* (1) If the input and/or output power consist of more than one component, such as multifrequency signals or noise, then the particular components used and their weighting must be specified. (2) This loss is usually expressed in decibels. *See:* **transducer.** 210

transducer, passive. A transducer that has no source of power other than the input signal(s), and whose output signal-power cannot exceed that of the input. *Note:* The definition of a passive transducer is a restriction of the more general **passive network,** that is, one containing no impressed driving forces. *See:* **transducer.** 210

transfer (1) (telephone switching systems). A feature that allows a customer to instruct the switching equipment or operator to transfer his call to another station. 55

(2) (electronic computation). (A) To transmit, or copy, information from one device to another. (B) To jump. (C) The act of transferring. *See:* **jump; transmit.** 235

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TENTH EDITION

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trans·at·lan·tic \ˌtran(t)s-ət-ˈlan-tik, ˌtranz-\ *adj* (1779) **1 a :** crossing or extending across the Atlantic Ocean ⟨a ~ cable⟩ **b :** relating to or involving crossing the Atlantic Ocean ⟨~ air fares⟩ **2 a :** situated or originating from beyond the Atlantic Ocean **b :** of, relating to, or involving countries on both sides of the Atlantic Ocean and esp. the U.S. and Great Britain ⟨~ cooperation⟩

trans·ax·le \tran(t)s-ˈak-səl, tranz-\ *n* [*trans*mission + *axle*] (1958) **:** a unit that consists of a combination of transmission and front axle used in front-wheel-drive automobiles

trans·ceiv·er \tran(t)-ˈsē-vər\ *n* [*trans*mitter + re*ceiver*] (1934) **:** a radio transmitter-receiver that uses many of the same components for both transmission and reception

tran·scend \tran(t)-ˈsend\ *vb* [ME, fr. L *transcendere* to climb across, transcend, fr. *trans-* + *scandere* to climb — more at SCAN] *vt* (14c) **1 a :** to rise above or go beyond the limits of **b :** to triumph over the negative or restrictive aspects of **:** OVERCOME **c :** to be prior to, beyond, and above (the universe or material existence) **2 :** to outstrip or outdo in some attribute, quality, or power ~ *vi* **:** to rise above or extend notably beyond ordinary limits ***syn*** see EXCEED

tran·scen·dence \-ˈsen-dən(t)s\ *n* (1601) **:** the quality or state of being transcendent

tran·scen·den·cy \-dən(t)-sē\ *n* (1615) **:** TRANSCENDENCE

tran·scen·dent \-dənt\ *adj* [L *transcendent-, transcendens,* prp. of *transcendere*] (1598) **1 a :** exceeding usual limits **:** SURPASSING **b :** extending or lying beyond the limits of ordinary experience **c** *in Kantian philosophy* **:** being beyond the limits of all possible experience and knowledge **2 :** being beyond comprehension **3 :** transcending the universe or material existence — **tran·scen·dent·ly** *adv*

tran·scen·den·tal \ˌtran(t)-ˌsen-ˈden-tᵊl, -sən-\ *adj* (1624) **1 a :** TRANSCENDENT 1b **b :** SUPERNATURAL **c :** ABSTRUSE, ABSTRACT **d :** of or relating to transcendentalism **2 a :** incapable of being the root of an algebraic equation with rational coefficients ⟨π is a ~ number⟩ **b :** being, involving, or representing a function (as sin *x*, log *x*, e^x) that cannot be expressed by a finite number of algebraic operations ⟨~ curves⟩ **3** *in Kantian philosophy* **a :** of or relating to experience as determined by the mind's makeup **b :** transcending experience but not human knowledge **4 :** TRANSCENDENT 1a — **tran·scen·den·tal·ly** \-tᵊl-ē\ *adv*

tran·scen·den·tal·ism \-tᵊl-ˌi-zəm\ *n* (1803) **1 :** a philosophy that emphasizes the a priori conditions of knowledge and experience or the unknowable character of ultimate reality or that emphasizes the transcendent as the fundamental reality **2 :** a philosophy that asserts the primacy of the spiritual and transcendental over the material and empirical **3 :** the quality or state of being transcendental; *esp* **:** visionary idealism — **tran·scen·den·tal·ist** \-tᵊl-ist\ *adj or n*

transcendental meditation *n* (1966) **:** a technique of meditation in which a mantra is chanted in order to foster calm, creativity, and spiritual well-being

trans·con·ti·nen·tal \ˌtran(t)s-ˌkän-tᵊn-ˈen-tᵊl, ˌtranz-\ *adj* (1853) **:** extending or going across a continent ⟨a ~ railroad⟩

tran·scribe \tran(t)-ˈskrīb\ *vt* **tran·scribed; tran·scrib·ing** [L *transcribere,* fr. *trans-* + *scribere* to write — more at SCRIBE] (1552) **1 a :** to make a written copy of **b :** to make a copy of (dictated or recorded matter) in longhand or on a machine (as a typewriter) **c :** to paraphrase or summarize in writing **d :** WRITE DOWN, RECORD **2 a :** to represent (speech sounds) by means of phonetic symbols **b :** TRANSLATE 2a **c :** to transfer (data) from one recording form to another **d :** to record (as on magnetic tape) for later broadcast **3 :** to make a musical transcription of **4 :** to cause (as DNA) to undergo genetic transcription — **tran·scrib·er** *n*

tran·script \ˈtran(t)-ˌskript\ *n* [ME, fr. ML *transcriptum,* fr. L, neut. of *transcriptus,* pp. of *transcribere*] (14c) **1 a :** a written, printed, or typed copy; *esp* **:** a usu. typewritten copy of dictated or recorded material **b :** an official or legal and often published copy ⟨a court reporter's ~⟩; *esp* **:** an official copy of a student's educational record **2 :** a representation (as of experience) in an art form **3 :** a sequence of RNA produced by transcription from a DNA template

tran·scrip·tase \tran-ˈskrip-ˌtās, -ˌtāz\ *n* [*transcrip*tion + *-ase*] (1963) **:** RNA POLYMERASE; *also* **:** REVERSE TRANSCRIPTASE

tran·scrip·tion \tran(t)-ˈskrip-shən\ *n* (1598) **1 :** an act, process, or instance of transcribing **2 :** COPY, TRANSCRIPT: as **a :** an arrangement of a musical composition for some instrument or voice other than the original **b :** a recording (as on magnetic tape) made esp. for use in radiobroadcasting **3 :** the process of constructing a messenger RNA molecule using a DNA molecule as a template with resulting transfer of genetic information to the messenger RNA — compare TRANSLATION 2, REVERSE TRANSCRIPTION — **tran·scrip·tion·al** \-shnəl, -shə-nᵊl\ *adj* — **tran·scrip·tion·al·ly** *adv*

tran·scrip·tion·ist \-shə-nist\ *n* (1963) **:** one that transcribes; *esp* **:** a typist who transcribes dictated medical reports

trans·cul·tur·al \tran(t)s-ˈkəl-chə-rəl, tranz-, -ˈkəlch-rəl\ *adj* (1951) **:** involving, encompassing, or extending across two or more cultures ⟨~ problems⟩

trans·cu·ta·ne·ous \ˌtran(t)s-kyu̇-ˈtā-nē-əs\ *adj* (ca. 1941) **:** passing, entering, or made by penetration through the skin ⟨~ infection⟩ ⟨~ inoculation⟩

trans·der·mal \tran(t)s-ˈdər-məl, tranz-\ *adj* (1944) **:** relating to, being, or supplying a medication in a form for absorption through the skin into the bloodstream ⟨~ drug delivery⟩ ⟨~ nitroglycerin⟩ ⟨~ nicotine patch⟩

trans·dis·ci·plin·ary \-ˈdi-sə-plə-ˌner-ē\ *adj* (1948) **:** INTERDISCIPLINARY

trans·duce \tran(t)s-ˈdüs, tranz-, -ˈdyüs\ *vt* **trans·duced; trans·duc·ing** [L *transducere* to lead across, transfer, fr. *trans-* + *ducere* to lead — more at TOW] (1947) **1 :** to convert (as energy or a message) into another form ⟨essentially sense organs ~ physical energy into a nervous signal⟩ **2 :** to bring about the transfer of (as a gene) from one microorganism to another by means of a viral agent

trans·duc·er \-ˈdü-sər, -ˈdyü-\ *n* (1924) **:** a device that is actuated by power from one system and supplies power usu. in another form to a second system ⟨a loudspeaker is a ~ that transforms electrical signals into sound energy⟩

trans·duc·tion \-ˈdək-shən\ *n* [L *transducere*] (1947) **:** the action or process of transducing; *esp* **:** the transfer of genetic determinants from one microorganism to another by a viral agent (as a bacteriophage) — **trans·duc·tant** \-tənt\ *n* — **trans·duc·tion·al** \-shnəl, -shə-nᵊl\ *adj*

1**tran·sect** \tran(t)-ˈsekt\ *vt* [*trans-* + inter*sect*] (1634) **:** to cut transversely — **tran·sec·tion** \-ˈsek-shən\ *n*

2**tran·sect** \ˈtran(t)-ˌsekt\ *n* (1905) **:** a sample area (as of vegetation) usu. in the form of a long continuous strip

tran·sept \ˈtran(t)-ˌsept\ *n* [NL *transeptum,* fr. L *trans-* + *septum, saeptum* enclosure, wall] (ca. 1542) **:** the part of a cruciform church that crosses at right angles to the greatest length between the nave and the apse or choir; *also* **:** either of the projecting ends of a transept — **tran·sep·tal** \tran(t)-ˈsep-tᵊl\ *adj*

trans·fec·tion \tran(t)s-ˈfek-shən, tranz-\ *n* [*trans-* + in*fection*] (1964) **:** infection of a cell with isolated viral nucleic acid followed by production of the complete virus in the cell; *also* **:** the incorporation of exogenous DNA into a cell — **trans·fect** \-ˈfekt\ *vt*

1**trans·fer** \tran(t)s-ˈfər, ˈtran(t)s-ˌ\ *vb* **trans·ferred; trans·fer·ring** [ME *transferren,* fr. L *transferre,* fr. *trans-* + *ferre* to carry — more at BEAR] *vt* (14c) **1 a :** to convey from one person, place, or situation to another **:** TRANSPORT **b :** to cause to pass from one to another **:** TRANSMIT **c :** TRANSFORM, CHANGE **2 :** to make over the possession or control of **:** CONVEY **3 :** to print or otherwise copy from one surface to another by contact ~ *vi* **1 :** to move to a different place, region, or situation; *esp* **:** to withdraw from one educational institution to enroll at another **2 :** to change from one vehicle or transportation line to another — **trans·fer·abil·i·ty** \(ˌ)tran(t)s-ˌfər-ə-ˈbi-lə-tē\ *n* — **trans·fer·able** *also* **trans·fer·ra·ble** \tran(t)s-ˈfər-ə-bəl\ *adj* — **trans·fer·al** \-əl\ *n* — **trans·fer·rer** \-ər\ *n*

2**trans·fer** \ˈtran(t)s-ˌfər\ *n* (1674) **1 a :** conveyance of right, title, or interest in real or personal property from one person to another **b :** removal or acquisition of property by mere delivery with intent to transfer title **2 a :** an act, process, or instance of transferring **:** TRANSFERENCE 2 **b :** the carryover or generalization of learned responses from one type of situation to another **3 :** one that transfers or is transferred; *esp* **:** a graphic image transferred by contact from one surface to another **4 :** a place where a transfer is made (as of trains to ferries or as where one form of power is changed to another) **5 :** a ticket entitling a passenger on a public conveyance to continue the trip on another route

trans·fer·ase \ˈtran(t)s-(ˌ)fər-ˌās, -ˌāz\ *n* (1948) **:** an enzyme that promotes transfer of a group from one molecule to another

trans·fer·ee \ˌtran(t)s-(ˌ)fər-ˈē\ *n* (ca. 1736) **1 :** a person to whom a conveyance is made **2 :** one who is transferred

trans·fer·ence \tran(t)s-ˈfər-ən(t)s, ˈtran(t)s-(ˌ)\ *n* (1681) **1 :** an act, process, or instance of transferring **:** CONVEYANCE, TRANSFER **2 :** the redirection of feelings and desires and esp. of those unconsciously retained from childhood toward a new object (as a psychoanalyst conducting therapy) — **trans·fer·en·tial** \ˌtran(t)s-fə-ˈren(t)-shəl\ *adj*

transfer factor *n* (1956) **:** a substance that is produced and secreted by a lymphocyte functioning in cell-mediated immunity and that upon incorporation into a lymphocyte which has not been sensitized confers on it the same immunological specificity as the sensitized cell

trans·fer·or \ˌtran(t)s-(ˌ)fər-ˈȯr\ *n* (1875) **:** one that conveys a title, right, or property

transfer payment *n* (ca. 1945) **1 :** a public expenditure made for a purpose (as unemployment compensation) other than procuring goods or services — usu. used in pl. **2** *pl* **:** money (as welfare payments) that is received by individuals and that is neither compensation for goods or services currently supplied nor income from investments

trans·fer·rin \tran(t)s-ˈfer-ən\ *n* [*trans-* + L *ferrum* iron] (1947) **:** a beta globulin in blood plasma capable of combining with ferric ions and transporting iron in the body

transfer RNA \ˈtran(t)s-ˌfər-\ *n* (1961) **:** a relatively small RNA that transfers a particular amino acid to a growing polypeptide chain at the ribosomal site of protein synthesis during translation — compare MESSENGER RNA

trans·fig·u·ra·tion \(ˌ)tran(t)s-ˌfi-gyə-ˈrā-shən, -gə-\ *n* (14c) **1 a :** a change in form or appearance **:** METAMORPHOSIS **b :** an exalting, glorifying, or spiritual change **2** *cap* **:** August 6 observed as a Christian feast in commemoration of the transfiguration of Christ on a mountaintop in the presence of three disciples

trans·fig·ure \tran(t)s-ˈfi-gyər, *esp Brit* -ˈfi-gər\ *vt* **-ured; -ur·ing** [ME, fr. L *transfigurare,* fr. *trans-* + *figurare* to shape, fashion, fr. *figura* figure] (14c) **:** to give a new and typically exalted or spiritual appearance to **:** transform outwardly and usu. for the better ***syn*** see TRANSFORM

trans·fi·nite \(ˌ)tran(t)s-ˈfī-ˌnīt\ *adj* [G *transfinit,* fr. *trans-* (fr. L) + *finit* finite, fr. L *finitus*] (1902) **1 :** going beyond or surpassing any finite number, group, or magnitude **2 :** being or relating to cardinal and ordinal numbers of sets with an infinite number of elements

trans·fix \tran(t)s-ˈfiks\ *vt* [L *transfixus,* pp. of *transfigere,* fr. *trans-* + *figere* to fasten, pierce — more at FIX] (1590) **1 :** to pierce through with or as if with a pointed weapon **:** IMPALE **2 :** to hold motionless by or as if by piercing — **trans·fix·ion** \-ˈfik-shən\ *n*

1**trans·form** \tran(t)s-ˈfȯrm\ *vb* [ME, fr. L *transformare,* fr. *trans-* + *formare* to form, fr. *forma* form] *vt* (14c) **1 a :** to change in composition or structure **b :** to change the outward form or appearance of **c :** to change in character or condition **:** CONVERT **2 :** to subject to mathematical transformation **3 :** to cause (a cell) to undergo genetic transformation ~ *vi* **:** to become transformed **:** CHANGE — **trans·form·able** \-ˈfȯr-mə-bəl\ *adj* — **trans·for·ma·tive** \-ˈfȯr-mə-tiv\ *adj*

syn TRANSFORM, METAMORPHOSE, TRANSMUTE, CONVERT, TRANSMOGRIFY, TRANSFIGURE mean to change a thing into a different thing. TRANSFORM implies a major change in form, nature, or function ⟨*transformed* a small company into a corporate giant⟩. METAMORPHOSE suggests an abrupt or startling change induced by or as if by magic or a supernatural power ⟨awkward girls *metamorphosed* into graceful ballerinas⟩. TRANSMUTE implies transforming into a higher element or thing ⟨at-

\ə\ abut \ᵊ\ kitten, F table \ər\ further \a\ ash \ā\ ace \ä\ mop, mar
\au̇\ out \ch\ chin \e\ bet \ē\ easy \g\ go \i\ hit \ī\ ice \j\ job
\ŋ\ sing \ō\ go \ȯ\ law \ȯi\ boy \th\ thin \t͟h\ the \ü\ loot \u̇\ foot
\y\ yet \zh\ vision \à, k̲, ⁿ, œ, œ̄, ue, u̅e, ʸ\ *see* Guide to Pronunciation

INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.992.3

(07/2002)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

Digital sections and digital line system – Access networks

Asymmetric digital subscriber line transceivers 2 (ADSL2)

ITU-T Recommendation G.992.3

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For further details, please refer to the list of ITU-T Recommendations.

2WIRE00030377

ITU-T Recommendation G.992.3

Asymmetric digital subscriber line transceivers 2 (ADSL2)

Summary

This Recommendation describes Asymmetric Digital Subscriber Line (ADSL) Transceivers on a metallic twisted pair that allows high-speed data transmission between the network operator end (ATU-C) and the customer end (ATU-R). It defines a variety of frame bearers in conjunction with one of two other services or without underlying service, dependent on the environment:

1) ADSL transmission simultaneously on the same pair with voice band service;
2) ADSL transmission simultaneously on the same pair with ISDN (Appendix I or II/G.961 [1]) services;
3) ADSL transmission without underlying service, optimized for deployment with ADSL over voiceband service in the same binder cable;
4) ADSL transmission without underlying service, optimized for deployment with ADSL over ISDN service in the same binder cable.

ADSL transmission on the same pair with voiceband services and operating in an environment with TCM-ISDN (Appendix III/G.961 [1]) services in an adjacent pair, is for further study.

This Recommendation specifies the physical layer characteristics of the Asymmetric Digital Subscriber Line (ADSL) interface to metallic loops.

This Recommendation has been written to help ensure the proper interfacing and interworking of ADSL transmission units at the customer end (ATU-R) and at the network operator end (ATU-C), and also to define the transport capability of the units. Proper operation shall be ensured when these two units are manufactured and provided independently. A single twisted pair of telephone wires is used to connect the ATU-C to the ATU-R. The ADSL transmission units must deal with a variety of wire pair characteristics and typical impairments (e.g., crosstalk and noise).

An ADSL transmission unit can simultaneously convey all of the following: a number of downstream frame bearers, a number of upstream frame bearers, a baseband POTS/ISDN duplex channel, and ADSL line overhead for framing, error control, operations, and maintenance. Systems support a net data rate ranging up to a minimum of 8 Mbit/s downstream and 800 kbit/s upstream. Support of net data rates above 8 Mbit/s downstream and support of net data rates above 800 kbit/s upstream are optional.

This Recommendation includes mandatory requirements, recommendations and options; these are designated by the words "shall", "should" and "may" respectively. The word "will" is used only to designate events that take place under some defined set of circumstances.

This Recommendation defines several optional capabilities and features:

- transport of STM and/or ATM and/or Packets;
- transport of a network timing reference;
- multiple latency paths;
- multiple frame bearers;
- short initialization procedure;
- dynamic rate repartitioning;
- seamless rate adaptation.

It is the intention of this Recommendation to provide, by negotiation during initialization, for U-interface compatibility and interoperability between transceivers complying with this Recommendation and between transceivers that include different combinations of options.

History

This Recommendation describes the second generation of ADSL, based on the first generation ITU-T Rec. G.992.1. It is intended that this Recommendation be implemented in multi-mode devices that support both ITU-T Recs G.992.3 and G.992.1.

This Recommendation has been written to provide additional features, relative to ITU-T Rec. G.992.1. ITU-T Rec. G.992.1 was approved in June 1999. Since then, several potential improvements have been identified in areas such as data rate versus loop reach performance, loop diagnostics, deployment from remote cabinets, spectrum control, power control, robustness against loop impairments and RFI, and operations and maintenance. This Recommendation provides a new ADSL U-interface specification, including the identified improvements, which the ITU-T believes will be most helpful to the ADSL industry.

Relative to ITU-T Rec. G.992.1, the following application-related features have been added:

- Improved application support for an all digital mode of operation and voice over ADSL operation;
- Packet TPS-TC function, in addition to the existing STM and ATM TPS-TC functions;
- Mandatory support of 8 Mbit/s downstream and 800 kbit/s upstream for TPS-TC function #0 and frame bearer #0;
- Support for IMA in the ATM TPS-TC;
- Improved configuration capability for each TPS-TC with configuration of latency, BER and minimum, maximum and reserved data rate.

Relative to ITU-T Rec. G.992.1, the following PMS-TC-related features have been added:

- A more flexible framing, including support for up to 4 frame bearers, 4 latency paths;
- Parameters allowing enhanced configuration of the overhead channel;
- Frame structure with receiver selected coding parameters;
- Frame structure with optimized use of RS coding gain;
- Frame structure with configurable latency and bit error ratio;
- OAM protocol to retrieve more detailed performance monitoring information;
- Enhanced on-line reconfiguration capabilities including dynamic rate repartitioning.

Relative to ITU-T Rec. G.992.1, the following PMD-related features have been added:

- New line diagnostics procedures available for both successful and unsuccessful initialization scenarios, loop characterization and troubleshooting;
- Enhanced on-line reconfiguration capabilities including bitswaps and seamless rate adaptation;
- Optional short initialization sequence for recovery from errors or fast resumption of operation;
- Optional seamless rate adaptation with line rate changes during showtime;
- Improved robustness against bridged taps with receiver determined pilot tone;
- Improved transceiver training with exchange of detailed transmit signal characteristics;
- Improved SNR measurement during channel analysis;
- Subcarrier blackout to allow RFI measurement during initialization and SHOWTIME;
- Improved performance with mandatory support of trellis coding;
- Improved performance with mandatory one-bit constellations;
- Improved performance with data modulated on the pilot tone;
- Improved RFI robustness with receiver determined tone ordering;
- Improved transmit power cutback possibilities at both CO and remote side;
- Improved Initialization with receiver and transmitter controlled duration of initialization states;
- Improved Initialization with receiver-determined carriers for modulation of messages;
- Improved channel identification capability with spectral shaping during Channel Discovery and Transceiver Training;
- Mandatory transmit power reduction to minimize excess margin under management layer control;
- Power saving feature for the central office ATU with new L2 low power state;
- Power saving feature with new L3 idle state;
- Spectrum control with individual tone masking under operator control through CO-MIB;
- Improved conformance testing including increase in data rates for many existing tests.

Through negotiation during initialization, the capability of equipment to support the G.992.3 and/or the G.992.1 Recommendations is identified. For reasons of interoperability, equipment may choose to support both Recommendations, such that it is able to adapt to the operating mode supported by the far-end equipment.

Source

ITU-T Recommendation G.992.3 was approved by ITU-T Study Group 15 (2001-2004) under the ITU-T Recommendation A.8 procedure on 29 July 2003.

It integrates the modifications introduced by ITU-T Rec. G.992.3 (2002) Amendment 1 approved on 22 May 2003.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g. interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

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ITU-T Recommendation G.992.3

Asymmetric digital subscriber line transceivers 2 (ADSL2)

1 Scope

For interrelationships of this Recommendation with other G.99x-series Recommendations, see ITU-T Rec. G.995.1 [B1].

This Recommendation describes the interface between the telecommunications network and the customer installation in terms of their interaction and electrical characteristics. The requirements of this Recommendation apply to a single asymmetric digital subscriber line (ADSL).

ADSL provides a variety of frame bearers in conjunction with other services:

- ADSL service on the same pair with voiceband services (including POTS and voiceband data services). The ADSL service occupies a frequency band above the voiceband service, and is separated from it by filtering;
- ADSL service on the same pair as ISDN service, as defined in Appendices I and II/G.961 [1]. The ADSL service occupies a frequency band above the ISDN service, and is separated from it by filtering;

ADSL also provides a variety of frame bearers without baseband services (i.e., POTS or ISDN) being present on the same pair:

- ADSL service on a pair, with improved spectral compatibility with ADSL over POTS present on an adjacent pair;
- ADSL service on a pair, with improved spectral compatibility with ADSL over ISDN present on an adjacent pair.

In the direction from the network operator to the customer premises (i.e., the downstream direction), the frame bearers provided may include low-speed frame bearers and high-speed frame bearers; in the other direction from the customer premises to the Central office (i.e., the upstream direction), only low-speed frame bearers are provided.

The transmission system is designed to operate on two-wire twisted metallic copper pairs with mixed gauges. This Recommendation is based on the use of copper pairs without loading coils, but bridged taps are acceptable in all but a few unusual situations.

Operation on the same pair with voiceband services (e.g., POTS and voiceband data services), and with TCM-ISDN service as defined in Appendix III/G.961 [1] on an adjacent pair, is for further study.

An overview of Digital Subscriber Line Transceivers can be found in ITU-T Rec. G.995.1 [B1].

Specifically, this Recommendation:

- defines the Transmission Protocol Specific Transmission Convergence Sub-layer for ATM, STM and Packet transport through the frame bearers provided;
- defines the combined options and ranges of the frame bearers provided;
- defines the line code and the spectral composition of the signals transmitted by both ATU-C and ATU-R;
- defines the initialization procedure for both the ATU-C and the ATU-R;
- specifies the transmit signals at both the ATU-C and ATU-R;
- describes the organization of transmitted and received data into frames;
- defines the functions of the OAM channel.

In separate annexes it also:

- describes the transmission technique used to support the simultaneous transport of voiceband services and frame bearers (ADSL over POTS, Annex A) on a single twisted-pair;
- describes the transmission technique used to support the simultaneous transport of ISDN services as defined in Appendices I and II/G.961 [1], and frame bearers (ADSL over ISDN, Annex B) on a single twisted-pair;
- describes the transmission technique used to support the transport of only frame bearers on a pair, with improved spectral compatibility with ADSL over POTS present on adjacent pair (All Digital Mode, Annex I);
- describes the transmission technique used to support the transport of only frame bearers on a pair, with improved spectral compatibility with ADSL over ISDN present on adjacent pair (All Digital Mode, Annex J).

This Recommendation defines the minimal set of requirements to provide satisfactory simultaneous transmission between the network and the customer interface of a variety of frame bearers and other services such as POTS or ISDN. The Recommendation permits network providers an expanded use of existing copper facilities. All required physical layer aspects to ensure compatibility between equipment in the network and equipment at a remote location are specified. Equipment may be implemented with additional functions and procedures.

2 References

The following ITU-T Recommendations, and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[1] ITU-T Recommendation G.961 (1993), *Digital transmission system on metallic local lines for ISDN basic rate access*.

[2] ITU-T Recommendation G.994.1 (2002), *Handshake procedures for digital subscriber line (DSL) transceivers*.

[3] ITU-T Recommendation G.996.1 (2001), *Test procedures for digital subscriber line (DSL) transceivers*.

[4] ITU-T Recommendation G.997.1 (1999), *Physical layer management for digital subscriber line (DSL) transceivers*.

[5] ISO 8601:2000, *Data elements and interchange formats – Information interchange – Representation of dates and times*.

[6] ITU-T Recommendation O.42 (1988), *Equipment to measure non-linear distortion using the 4-tone intermodulation method.*

For Annex B

[7] ETSI TS 102 080 V1.3.2 (2000), *Transmission and Multiplexing (TM); Integrated Services Digital Network (ISDN) basic rate access; Digital transmission on metallic local lines*.

For Annex E

[8] ETSI TS 101 952-1 V1.1.1 (2002), *Specification of ADSL splitters for European deployment*.

RS	Reed Solomon
RT	Remote Terminal
RX	Receiver
SEF	Severely Errored Frame
SM	Service Module
SNR	Signal-to-Noise Ratio
TC	Transmission convergence (sublayer)
TP	Twisted Pair
TPS-TC	Transmission Protocol Specific TC Layer
T-R	Interface(s) between ATU-R and switching layer (ATM or STM or Packet)
T/S	Interface(s) between ADSL network termination and CPE or home network
TX	Transmitter
U-C	Loop Interface – Central Office end
U-R	Loop Interface – Remote Terminal end
UTC	Unable to comply
V-C	Logical interface between ATU-C and a digital network element such as one or more switching systems
ZHP	Impedance high-pass filter
4-QAM	4 point QAM (i.e., two bits per symbol)
$\oplus$	Exclusive-or; modulo-2 addition
$\lceil x \rceil$	Rounding to the higher integer

5 Reference models

G.992.3 devices fit within the family of DSL Recommendations described in ITU-T Rec. G.995.1 [B1]. Additionally, G.992.3 devices rely upon constituent components described within ITU-T Rec. G.994.1 [2] and ITU-T Rec. G.997.1 [4]. This clause provides the necessary functional, application, and protocol reference models so that the subclauses of this Recommendation may be related to these additional Recommendations.

5.1 ATU functional model

Figure 5-1 shows the functional blocks and interfaces of an ATU-C and ATU-R that are referenced in this Recommendation. It illustrates the most basic functionality of the ATU-R and the ATU-C. Each ATU contains both an application invariant section and an application specific section. The application invariant section consists of the PMS-TC and PMD layers and are defined in clauses 7 and 8, while the application specific aspects that are confined to the TPS-TC layer and device interfaces, are defined in Annex K. Management functions, which are typically controlled by the operator's management system (EMS or NMS), are not shown in the Figure 5-1. Figure 5-3 provides a high level view that includes the management interface.

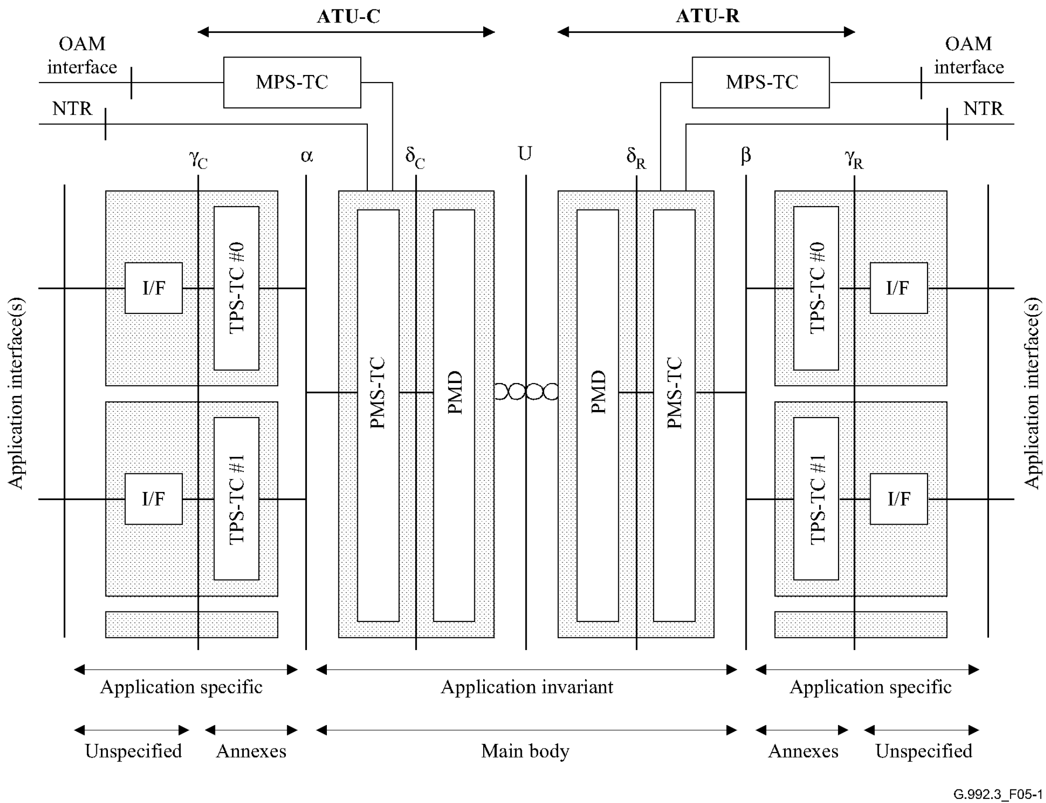

Figure 5-1/G.992.3 – ATU functional model

The principal functions of the PMD layer may include symbol timing generation and recovery, encoding and decoding, modulation and demodulation, echo cancellation (if implemented) and line equalization, link startup, and physical layer overhead (superframing). Additionally, the PMD layer may generate or receive control messages via the overhead channel of the PMS-TC layer.

The PMS-TC layer contains the framing and frame synchronization functions, as well as forward error correction, error detection, scrambler and descrambler functions. Additionally, the PMS-TC layer provides an overhead channel that is used to transport control messages generated in the TPS-TC, PMS-TC or PMD layers as well as messages generated at the management interface.

The PMS-TC is connected across the α and β interfaces in the ATU-C and the ATU-R, respectively, to the TPS-TC layer. The TPS-TC is application specific and consists largely of adaptation of the customer interface data and control signals to the (a)synchronous data interface of the TPS-TC. Additionally, the TPC-TC layer may also generate or receive control messages via the overhead channel of the PMS-TC layer.

The TPS-TC layer communicates with the interface blocks across the γ_R and γ_C interfaces. Depending upon the specific application, the TPS-TC layer may be required to support one or more channels of user data and associated interfaces. The definition of these interfaces is beyond the scope of this Recommendation.

The MPS-TC function provides procedures to facilitate the management of the ATU. The MPS-TC function communicates with higher layer functions in the management plane that are described in ITU-T Rec. G.997.1 [4] (e.g., the Element Management System, controlling the CO-MIB). Management information is exchanged between the MPS-TC functions through an

ADSL overhead channel. The PMS-TC multiplexes the ADSL overhead channel with the TPS-TC data streams for transmission over the DSL. The management information contains indications of anomalies and defects and related performance monitoring counters. In addition, several management command procedures are defined for use by higher layer functions, specifically for testing purposes.

The α, β, γ_R and γ_C interfaces are only intended as logical separations and need not be physically accessible. The γ_R and γ_C interfaces are logically equivalent to respectively the T-R and V-C interfaces shown in Figure 5-4.

5.2 User plane protocol reference model

The User Plane Protocol Reference Model, shown in Figure 5-2, is an alternate representation of the information shown in Figure 5-1. The user plane protocol reference model is included to emphasize the layered nature of this Recommendation and to provide a view that is consistent with the generic xDSL models shown in ITU-T Rec. G.995.1 [B1].

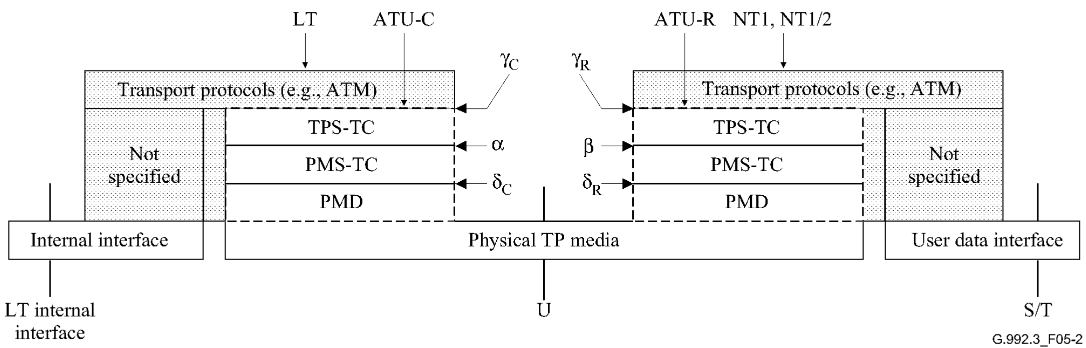

Figure 5-2/G.992.3 – User plane protocol reference model

The one way payload transfer delay between the γ_C and γ_R reference points is the sum of:

- Delay through the TPS-TC at ATU-C and ATU-R;
- Delay through the PMS-TC at ATU-C and ATU-R;
- Delay through the PMD at ATU-C and ATU-R.

The delay through the TPS-TC depends on the TPS-TC type used. The delay through the PMS-TC and PMD sublayer (i.e., the delay between the α and β reference points) can be modelled independently of the TPS-TC type used, and is referred to as the nominal one-way maximum payload transfer delay. It is defined as:

$$delay_{\alpha-\beta} = 3.75 + \frac{\lceil S_P \times D_P \rceil}{4} ms$$

where the $\lceil x \rceil$ notation denotes rounding to the higher integer,

and S_P and D_P are PMS-TC control parameters defined in 7.5 and 7.6.

Table 5-1 illustrates the data rate terminology and definitions as applicable at various reference points. The reference points refer to those shown in the reference model in Figure 5-2 and the PMS-TC block diagram in Figure 7-6.

Table 5-1/G.992.3 – Data rate terminology and definitions

Data rate	Equation (kbit/s)	Reference point
Net data rate	$\sum \text{Net}_{\text{p.act}}$ (see Table 7-7)	α, β
Aggregate data rate = Net data rate + Frame overhead rate	$\sum (\text{Net}_{\text{p.act}} + \text{OR}_{\text{P}})$ (see Table 7-7)	A
Total data rate = Aggregatedata rate + RS Coding overhead rate	$(\sum \text{L}_{\text{P}}) \times 4$ (see Table 7-6)	B, C, δ
Line rate = Total data rate + Trellis Coding overhead rate	$(\sum b_i) \times 4$ (see Table 8-4)	U

5.3 Management plane reference model

The Management Plane Protocol Reference Model, shown in Figure 5-3 is an alternate representation of the information shown in Figure 5-1. The management plane protocol reference model is included to emphasize the separate functions provided by the MPS-TC and TPS-TC functions and to provide a view that is consistent with the generic xDSL models shown in ITU-T Rec. G.995.1 [B1].

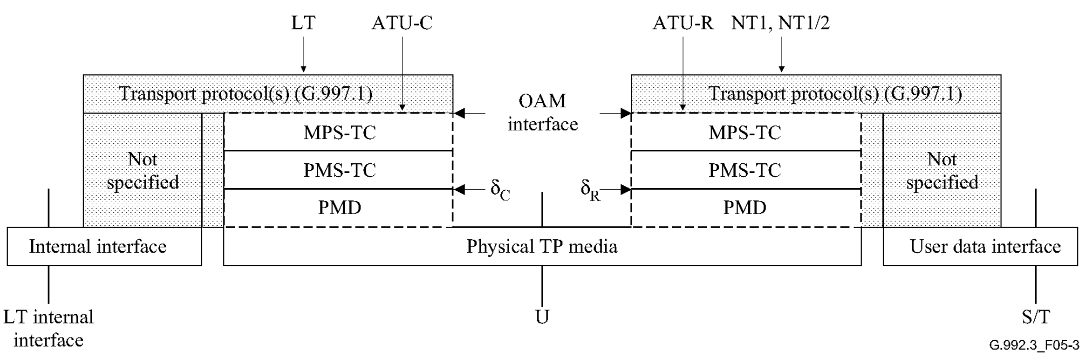

Figure 5-3/G.992.3 – Management plane protocol reference model

5.4 Application models

The application models for G.992.3 is based upon the generic reference configuration described in 6.1/G.995.1 [B1]. There are four separate applications models, one each for ADSL data service only, ADSL data service with underlying POTS service, ADSL data service with underlying ISDN service and Voice over ADSL service.

Two generic application models for G.992.3 exist. The application model for remote deployment with splitter is shown in Figure 5-4.

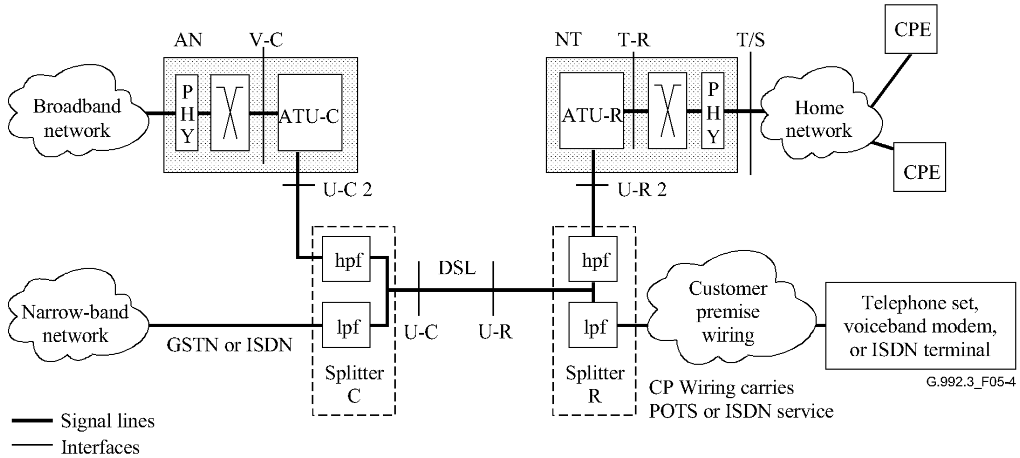

Figure 5-4/G.992.3 – Generic application reference model for remote deployment with splitter

The application model for splitterless remote deployment is shown in Figure 5-5. An optional low-pass filter may be included to provide isolation and protection of telephone sets, voiceband modems, ISDN terminals, and the ATU-R. The location of filters in all application model diagrams is intended to be functional only. The specific functions of the filter may be regionally specific. The filter may be implemented in a variety of ways, including splitters, in-line filters, integrated filters with ATU devices, and integrated filters with voice equipment.

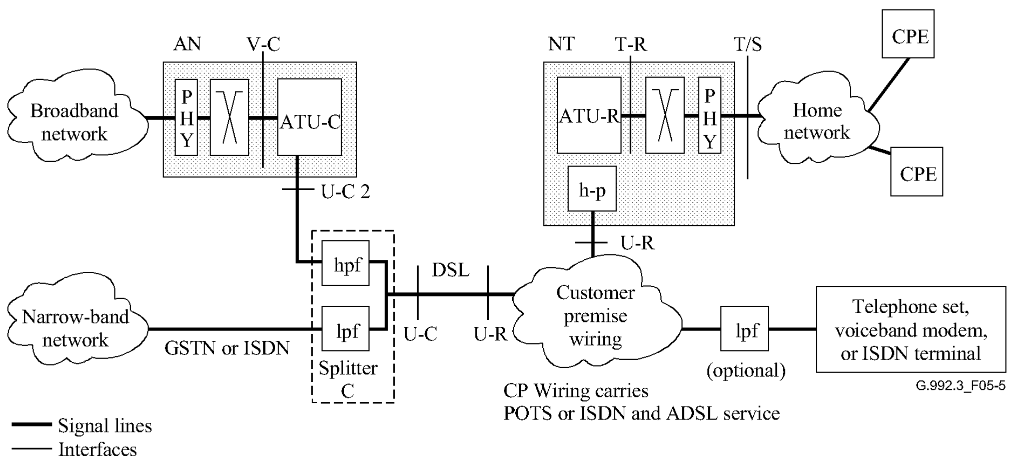

Figure 5-5/G.992.3 – Generic application reference model for splitterless remote deployment

NOTE 1 – The U-C and U-R interfaces are fully defined in this Recommendation. The V-C and T-R interfaces are defined only in terms of logical functions, not physical. The T/S interface is not defined in this Recommendation.

NOTE 2 – Implementation of the V-C and T-R interfaces is optional when interfacing elements are integrated into a common element.

NOTE 3 – One or other of the high-pass filters, which are part of the splitters, may be integrated into the ATU-x; if so, then the U-C 2 and U-R 2 interfaces become the same as the U-C and U-R interfaces, respectively.

NOTE 4 – More than one type of T-R interface may be defined, and more than one type of T/S interface may be provided from an ADSL NT (e.g., NT1 or NT2 types of functionalities).

NOTE 5 – A future issue of this Recommendation may deal with customer installation distribution and home network requirements.

NOTE 6 – Specifications for the splitters are given in Annex E.

5.4.1 Data service

Figure 5-6 depicts the typical application model for delivering data service over G.992.3, showing reference points and attached equipment. In such an application, an ATU-R is part of the ADSL NT which will typically connect to one or more user terminals, which may include data terminals, telecommunications equipment, or other devices. These connections to these pieces of terminal equipment are designated S/T reference points. The connection between ATU-R and ATU-C will typically be a direct one through a DSL, with the customer premises endpoint of the DSL designated as U-R reference point and the network endpoint designated U-C reference point. The ATU-C is part of the Access Node, which will typically connect to a broadband access network at the V reference point. In this application model there will be no associated narrowband service carried on the same DSL.

The ADSL may be operated in all digital mode, without underlying service, or, may be operated in the mode for underlying POTS or ISDN service, with the bandwidth reserved for the underlying service being unused.

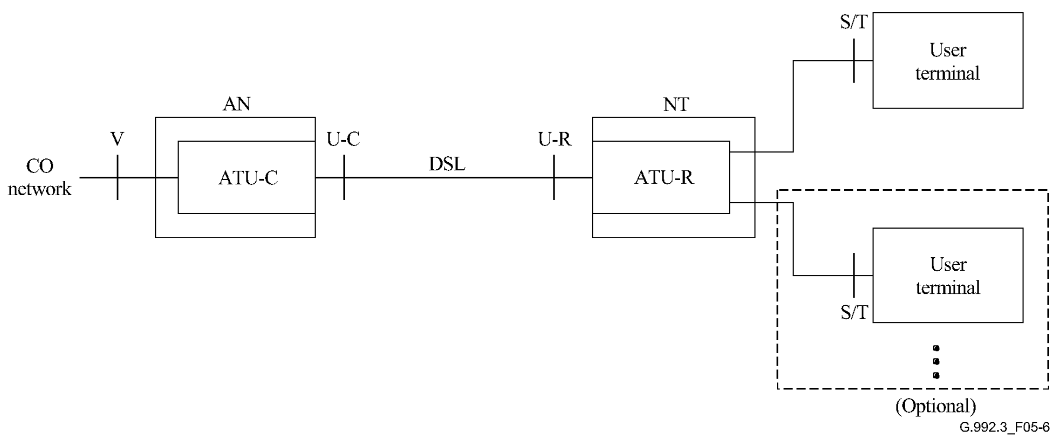

Figure 5-6/G.992.3 – Data service application model

5.4.2 Data with POTS service

Figure 5-7 depicts the typical application model for delivering data service over G.992.3 with an underlying POTS service on the same DSL, showing reference points and attached equipment. In such an application, an ATU-R is part of the ADSL NT which will typically connect to one or more user terminals, which may include data terminals, telecommunications equipment, or other devices. The connections to these pieces of terminal equipment are designated S/T reference points. The ATU-R will not be directly attached to the U-R reference point but will be separated from the DSL by a high- pass filter element. Additionally, one or more voice terminals will also be part of the application model at the customer premises. These voice terminals may include POTS telephones, telephone answering devices, voiceband analog modems, or other devices. The voice terminals may

be attached directly the U-R reference point or may be connected through a low-pass filter element per voice terminal (splitterless remote deployment) or may be connected through a common low-pass filter element (remote deployment with splitter). At the central endpoint of the DSL, the ATU-C will connect to the U-C reference point through a high-pass filter element. The ATU-C is part of the Access Node, which will typically connect to a broadband access network at the V reference point. Additionally, there will be a low-pass filter element attached at the U-C reference point to connect with the GSTN core network.

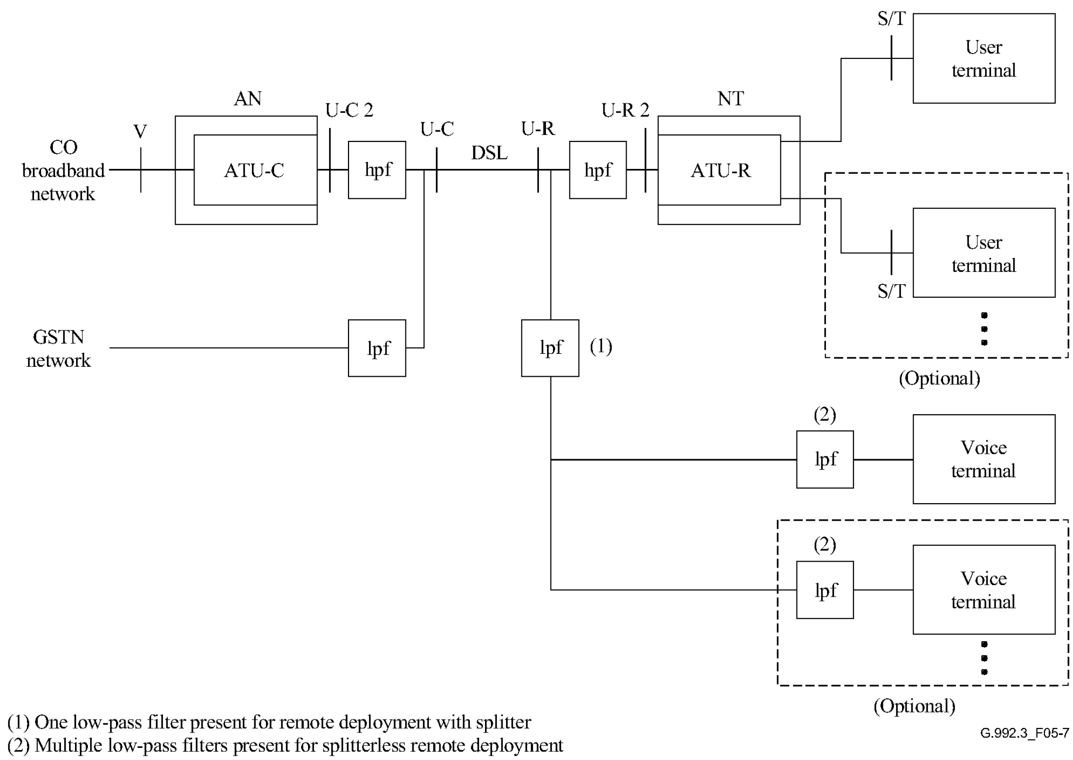

Figure 5-7/G.992.3 – Data with POTS service application model

NOTE – The low-pass filter shown at the customer premises in Figures 5-5 and 5-7 is also known as an in-line filter. The specification of in-line filter characteristics is outside the scope of this Recommendation. However, in-line filters are specified by regional standards bodies, e.g., see [B10].

5.4.3 Data with ISDN service

Figure 5-8 depicts the typical application model for delivering data service over G.992.3 with an underlying ISDN service on the same DSL, showing reference points and attached equipment. In such an application, the ATU-R is part of the ADSL NT which will typically connect to one or more user terminals which may include data terminals, telecommunications equipment, or other devices. The connections to these pieces of terminal equipment are designated S/T reference points. The ATU-R will not be directly attached to the U-R reference point but will be separated from the DSL by a high-pass filter element. One ISDN NT will also be part of the application model at the customer premises. The ISDN NT is not attached directly the U-R reference point but will be separated from the DSL by a low-pass filter element. Additionally, one or more voice terminals will also be part of the application model at the customer premises. These voice terminals are connected to the ISDN NT and may include POTS or ISDN telephones, telephone answering devices, voiceband analog modems, or other devices. At the central endpoint of the DSL, the ATU-C will

connect to the U-C reference point through a high-pass filter element. The ATU-C is part of the Access Node, which will typically connect to a broadband access network at the V reference point. Additionally, there will be a low-pass filter element attached at the U-C reference point to connect with the GSTN core network.

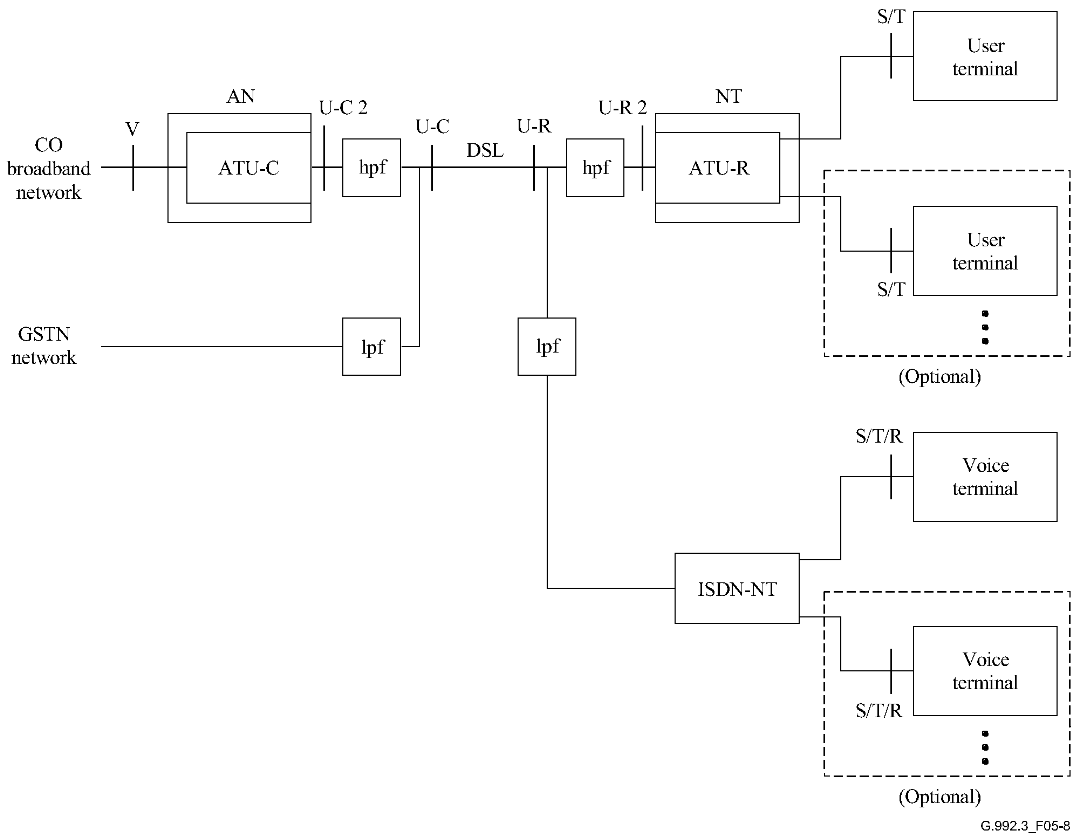

Figure 5-8/G.992.3 – Data with ISDN service application model

5.4.4 Voice over data service

Figure 5-9 depicts the typical application model for delivering data and voice services over G.992.3, showing reference points and attached equipment. In such an application, an ATU-R is part of the ADSL NT which will typically connect to one or more user terminals and to one or more voice terminals. The data terminals may include data terminals, telecommunications equipment, or other devices. The voice terminals may include POTS or ISDN telephone devices, telephone answering devices, voiceband analog modems, or other devices. The connections to these pieces of terminal equipment are designated S/T reference points. The ATU-R and ATU-C will include a voice interworking function that allows a connection from the GSTN network to the voice terminal equipment. The connection between ATU-R and ATU-C will typically be a direct one through a DSL, with the customer premises endpoint of the DSL designated as U-R reference point and the network endpoint designated U-C reference point. The ATU-C is part of the Access Node, which will typically connect to a broadband access network at the V reference point. In addition, the ATU-C will connect to the GSTN core network.

The ADSL may be operated in all digital mode, without underlying service, or, may be operated in the mode for underlying POTS or ISDN service, with the bandwidth reserved for the underlying service being unused, or, although not depicted in Figure 5-8, there may also be an underlying POTS or ISDN service delivered through the DSL.

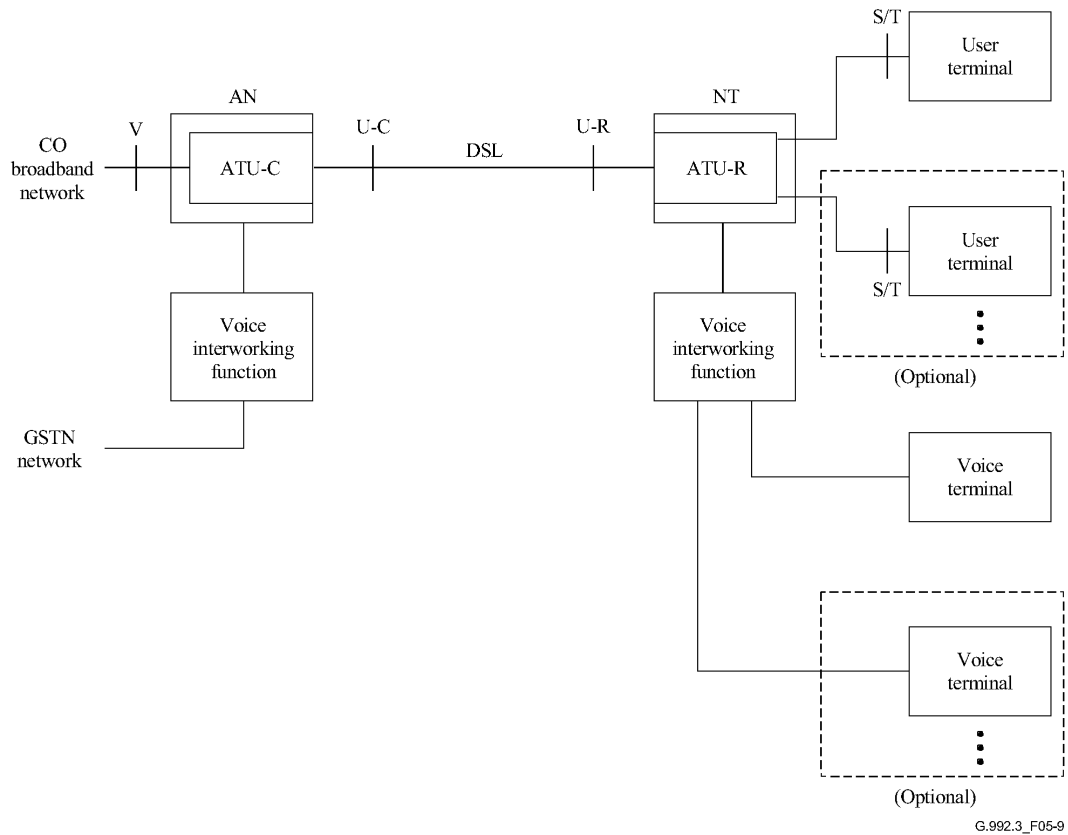

Figure 5-9/G.992.3 – Voice over data service application model

6 Transport Protocol Specific Transmission Convergence (TPS-TC) function

6.1 Transport capabilities

This Recommendation provides procedures for the transport of the output frame bearers of one to four unidirectional TPS-TC functions in both the upstream and downstream directions. For purposes of reference and identification, each of the TPS-TC functions within an ATU is labelled as if it were mapped to a particular frame bearer, i.e., TPS-TC #0, #1, #2, #3 would be mapped on frame bearer #0, #1, #2, #3 respectively. The TPS-TC functions may be of differing types, and each type is described in detail in Annex K.

After each of the transmit and receive TPS-TC functions has been mapped to a frame bearer during the G.994.1 phase of initialization, transport of the TPS-TC functions on frame bearers is carried out by underlying PMS-TC and PMD layers through a series of data frames and PMD symbols. The TPS-TC transport capabilities are configured by the control parameters described in Annex K. The control parameters provide for the application of appropriate data rates and characteristics of each TPS-TC function as if it were mapped to a particular frame bearer. Any receive TPS-TC function can be logically connected to any transmit TPS-TC function that supports the same TPS-TC

Table 7-2/G.992.3 – Signalling primitives to transport control messages over the pair of PMS-TC functions

Signal	Primitive	Description
Frame.Control	.request	The MPS-TC function uses this primitive to pass one entire control message for transport to the transmit PMS-TC function. Upon receipt of a message, the PMS-TC function shall begin the Transmitter Protocol procedure in 7.8.2.4.1.
	.confirm	This primitive is used by the transmit PMS-TC function to confirm receipt of a Frame.Control.request primitive. By the interworking of the request and confirm, the data flow is synchronized to the rate that can be accommodated by the overhead rate of the PMS-TC functions.
	.indicate	The receive PMS-TC function uses this primitive to pass a single control messages or indications that are received to the MPS-TC function.

Table 7-3/G.992.3 – Signalling primitives to transport NTR information over the pair of PMS-TC functions

Signal	Primitive	Description
Frame.NTR	.indicate	This primitive is used to convey the current phase of the NTR signal to the transmit PMS-TC function. Upon receipt of this primitive, the PMS-TC transmit function shall execute the NTR Transport procedure in 7.8.1. At the ATU-R, this primitive is passed by the receive PMS-TC function.

Table 7-4/G.992.3 – Signalling primitives to convey maintenance indications to the local maintenance entity

Signal	Primitive	Description
Management.Prim	.indicate	This primitive is used by various local functions within the ATU to pass management anomalies, defects and parameters to the transmit MPS-TC function. Upon receipt of this primitive, the transmit PMS-TC function shall execute the Indicator Bits procedure in 7.8.2.2. This primitive is used by the receive PMS-TC function to signal a number of anomaly supervisory primitives to the MPS-TC function.

7.4 Block diagram and internal reference point signals

Figure 7-6 depicts the functions within a transmit PMS-TC function that supports N_{BC} frame bearers ($1 \leq N_{BC} \leq 4$). These frame bearers (i.e., Frame.Bearer(*n*).confirm primitives from the transmit TPS-TC function) are shown at the leftmost edge of Figure 7-6. Within the transmit PMS-TC function, there are one to four latency path functions that accept input from zero, one, or more of the frame bearers. Within each latency path function, there are three reference points labeled A, B, and C. The output signals from each latency path function at Reference Point C are combined by an additional multiplexing function to form the PMD bits (i.e., PMD.Bits.confirm primitives to the transmit PMD function), depicted at the rightmost edge of Figure 7-6.

The control input signals are depicted at the uppermost edge of Figure 7-6. These are encoded onto a shared overhead channel, one octet associated with each of the latency path functions. These sync octets are combined with frame bearer data within the latency path function at Reference Point A.

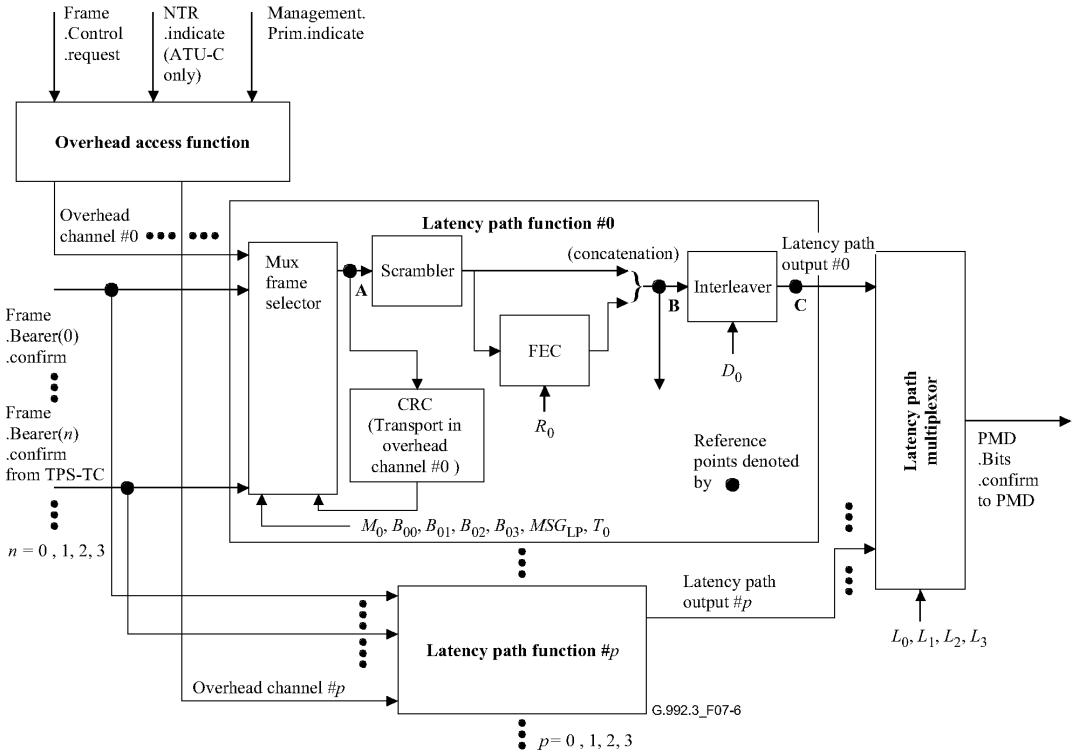

Figure 7-6/G.992.3 – Block diagram of transmit PMS-TC function

Because of the various functions depicted in Figure 7-6, the data within the transmit PMS-TC function has a different structural grouping as it moves from the frame bearers to the PMD bits. Reference points are defined within the block diagram for purposes of helping to depict this structure. These reference points are for clarity only. The reference points with which the PMS-TC procedures will be described are depicted in Figure 7-6 and listed in Table 7-5. It is important to note that all octet boundaries and positions of most significant bits in the frame bearers will be maintained at each of the reference points listed in Table 7-5.

Table 7-5/G.992.3 – PMS-TC function internal reference points

Reference point	Definition
A: Mux Data Frame	The data within a latency path function after the sync octet has been added.
B: FEC Data Frame	The data within a latency path function after the output of the FEC redundancy octets are merged with scrambled data.
C: Interleaved FEC Data Frame	The data and redundancy octets that have been interleaved. This is the output signal of a latency path function.

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7.5 Control parameters

The configuration of the PMS-TC function is controlled by a set of control parameters displayed in Table 7-6.

Table 7-6/G.992.3 – Framing Parameters

Parameter	Definition
MSG_{min}	The minimum rate of the message based overhead that shall be maintained by the ATU. MSG_{min} is expressed in bits per second.
MSG_{max}	The maximum rate of the message based overhead that shall be allowed by the ATU. MSG_{max} is expressed in bits per second.
N_{BC}	See Table 6-1. This is a TPS-TC configuration parameter repeated here for clarity.
N_{LP}	The number of latency paths enabled to transport frame bearers and overhead. The latency path functions are labeled #0, #1, #2 and #3.
MSG_{LP}	The label of the latency path used to transport the message based overhead information.
MSG_C	The number of octets in the message based portion of the overhead structure.
$B_{p.n}$	The nominal number of octets from frame bearer #*n* per Mux Data Frame at Reference Point A in latency path function #*p*. When T_p is not set to 1 and *n* is the lowest index of the frame bearers assigned to latency path #*p*, the number of octets from the frame bearer #*n* in the latency path function #*p* varies between $B_{p.n}$ and $B_{p.n} + 1$.
M_p	The number of Mux Data Frames per FEC Data Frame in latency path function #*p*.
T_p	The ratio of the number of Mux Data Frames to the number of sync octets in the latency path function #*p*. A sync octet is inserted with every T_p-th Mux Data Frame. When T_p is not set to one, an extra frame bearer octet is carried whenever a sync octet is not inserted.
R_p	The number of RS redundancy octets per codeword in latency path function #*p*. This is also the number of redundancy octet per FEC Data Frame in the latency path function #*p*.
D_p	The interleaving depth in the latency path function #*p*.
L_p	The number of bits from the latency path function #*p* included per PMD.Bits.confirm primitive.

The first two control parameters listed in Table 7-6 establish persistent constraints upon the operation of the PMS-TC function that apply during all initialization and reconfiguration procedures. The values of these control parameters shall be set during the G.994.1 phase of initialization, in accordance with common requirements of the ATU devices. The requirements for these control parameters by each ATU in each direction may also be exchanged during the G.994.1 phase of initialization.

The remaining control parameters listed in Table 7-6 establish the specific parameters that control the PMS-TC procedures described in this clause. The values of these control parameters shall be set during the PMD initialization procedure in accordance with capabilities of each ATU and requirements of each ATU's higher layers as determined by TPS-TC initialization procedures. Additionally, some of the control parameters in Table 7-6 may be modified during on-line reconfiguration procedures.

All valid control parameter configurations are described in 7.6.2. All mandatory control parameter configurations described in 7.6.3 shall be supported by each ATU.

7.6 Frame structure

The various transported data can be assigned various structural groupings as it moves through the transmit PMS-TC function. These taken together are termed the frame structure. The frame structure is defined for clarity only and the actual groupings within an ATU implementation may vary.

The ATU frame structure for the case of two frame bearers transported over a single latency path ($N_{BC} = 2$, $N_{LP} = 1$, $T_p = 1$) is illustrated in Figure 7-7. This figure shows the frame structure and data groupings at the start of the PMS-TC procedure, at each Reference Point A, B, and C of latency path function #0, and at the end of the PMS-TC procedure.

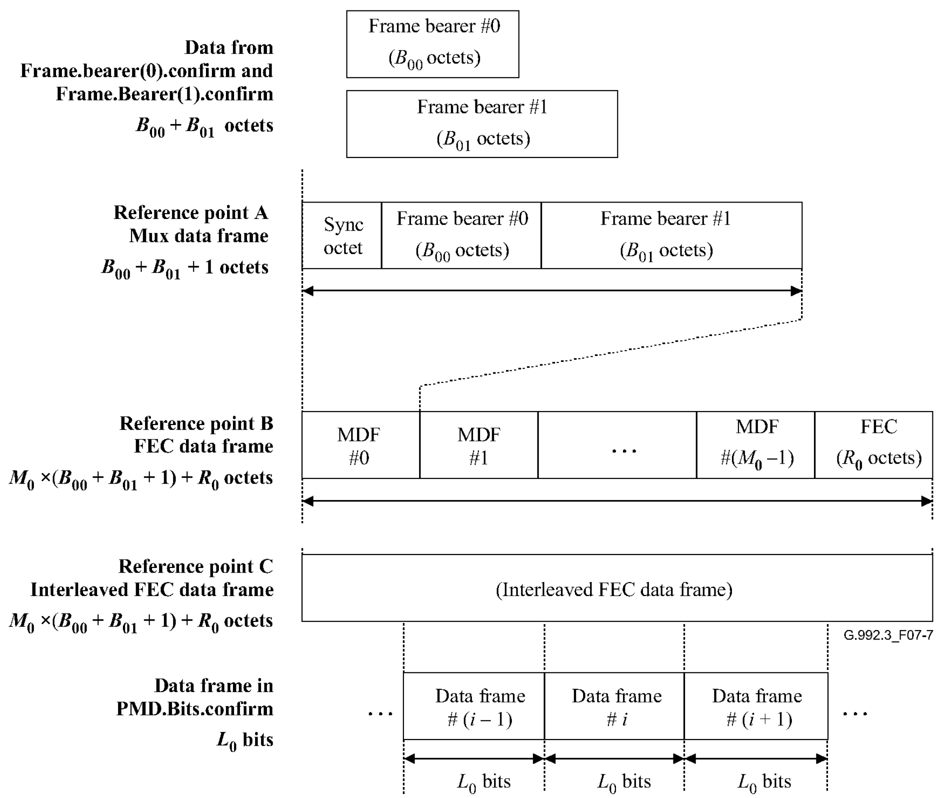

Figure 7-7/G.992.3 – Illustration of frame structure with single latency dual bearers and $T_p = 1$

As a further illustration, Figure 7-8 depicts the frame structure when the PMS-TC function is configured to support two frame bearers with two latency paths ($N_{BC} = 2$, $N_{LP} = 2$, $B_{00} = 0$, $B_{11} = 0$). MSG_{LP} is set to one and $T_0 = 1$. Figure 7-8 illustrates PMS-TC functions for a Mux Data Frame (MDF) that does not include the sync octet for the second latency, assuming that T_1 is not set to 1 for this example and the current mux data frame selector counter modulo T_p is not equal to 0.

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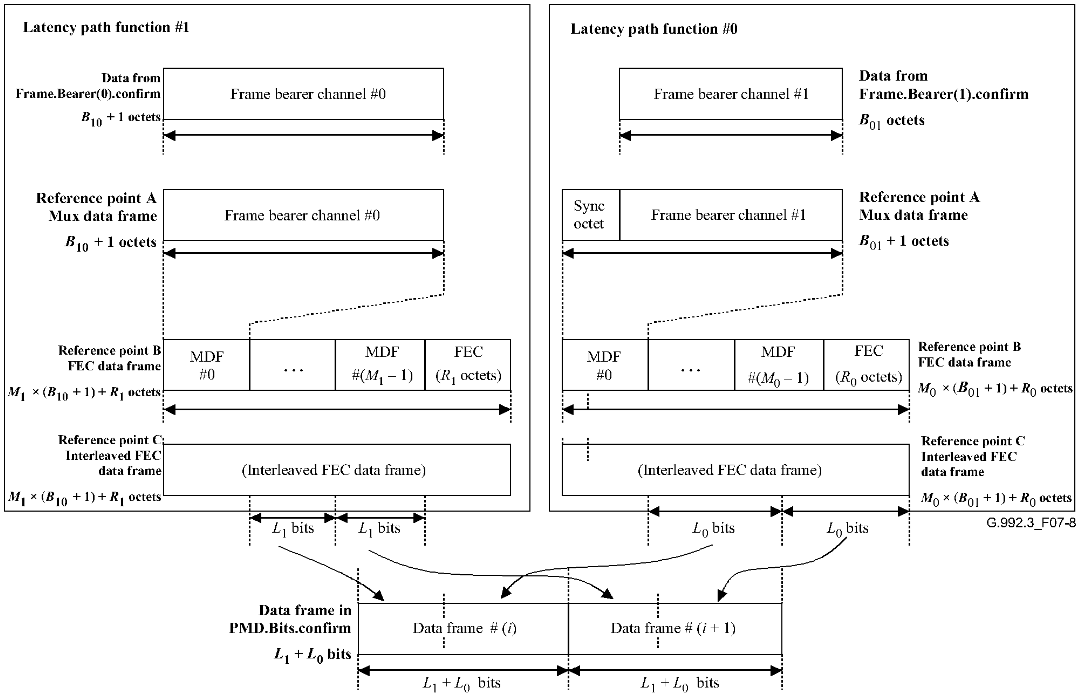

Figure 7-8/G.992.3 – Illustration of frame structure with dual latency and dual bearers

7.6.1 Derived definitions

Table 7-7 displays several definitions of symbols that derive from the PMS-TC control parameters and that are used to describe characteristics of the ATU data frame. These definitions are for clarity only.

Table 7-7/G.992.3 – Derived characteristics of the ATU data frame

Symbols	Definition and value
K_p	**Definition**: The number of octets per Mux Data Frame in latency path function #p $K_p = \sum_{i=0}^{N_{BC}-1} B_{p,i} + 1$
$N_{FEC.p}$	**Definition**: The number of octets per FEC Data Frame and Interleaved FEC Data Frame in latency path function #p $N_{FEC.p} = M_p \times K_p + R_p$
S_p	**Definition**: The number of PMD.Bits.request primitives (and correspondingly the number of PMD symbols) over which the FEC Data Frame spans, not accounting for the $S_p = \frac{8 \times N_{FEC.p}}{L_p}$ The value of S_p may represent a non-integer value.

Table 7-7/G.992.3 – Derived characteristics of the ATU data frame

Symbols	Definition and value
$net\_act_{p.n}$	**Definition**: Net data rate of frame bearer #n in latency path function #p When $T_p = 1$: $net\_act_{p.n} = \frac{B_{p.n} \times M_p}{S_p} \times 32 \text{ kbit/s} = \frac{B_{p.n} \times M_p \times L_p}{K_p \times M_p + R_p} \times 4 \text{ kbit/s}$ When $T_p \neq 1$, for bearers associated to the lowest index: $net\_act_{p.n} = \left(\frac{B_{p.n} \times M_p}{S_p} + \frac{(T_p - 1) \times M_p}{T_p \times S_p} \right) \times 32 \text{ kbit/s}$ $= \frac{(T_p \times (B_{p.n} + 1) - 1) \times M_p \times L_p}{T_p \times (K_p \times M_p + R_p)} \times 4 \text{ kbit/s}$ for bearers associated with subsequence values in the list: $net\_act_{p.n} = \frac{B_{p.n} \times M_p}{S_p} \times 32 \text{ kbit/s} = \frac{B_{p.n} \times M_p \times L_p}{K_p \times M_p + R_p} \times 4 \text{ kbit/s}$
$Net_{p.act}$	**Definition**: Net data rate of latency path function #p When $T_p = 1$, $Net_{p.act} = \frac{(K_p - 1) \times M_p}{S_p} \times 32 \text{ kbit/s} = \frac{(K_p - 1) \times M_p \times L_p}{K_p \times M_p + R_p} \times 4 \text{ kbit/s}$ When $T_p \neq 1$, $Net_{p.act} = \left(\frac{(K_p - 1) \times M_p}{S_p} + \frac{(T_p - 1) \times M_p}{T_p \times S_p} \right) \times 32 \text{ kbit/s}$ $= \frac{(T_p \times K_p - 1) \times M_p \times L_p}{T_p \times (K_p \times M_p + R_p)} \times 4 \text{ kbit/s}$
OR_p	**Definition**: Overhead rate of latency path function #p $OR_p = \frac{M_p}{T_p \times S_p} \times 32 \text{ kbit/s} = \frac{M_p \times L_p}{T_p \times (K_p \times M_p + R_p)} \times 4 \text{ kbit/s}$
$delay_p$	**Definition**: PMS-TC delay of latency path function #p Nominal one-way maximum transport delay of latency path function #p is defined as: $delay_p = \frac{\lceil S_p \times D_p \rceil}{4} ms, (\text{where } \lceil x \rceil \text{ denotes rounding to the higher integer})$
SEQ_p	**Definition**: Length of the sync octet sequence of latency path function #p $SEQ_p = \begin{cases} 2 & \text{if } p \neq MSG_{LP} \text{ and latency path } \# p \text{ is not the lowest latency path (See 7.8.2.1)} \\ 6 & \text{if } p \neq MSG_{LP} \text{ and latency path } \# p \text{ is the lowest latency path (See 7.8.2.1)} \\ MSG_C + 2 & \text{if } p = MSG_{LP} \text{ and latency path } \# p \text{ is not the lowest latency path (See 7.8.2.1)} \\ MSG_C + 6 & \text{if } p = MSG_{LP} \text{ and latency path } \# p \text{ is the lowest latency path (See 7.8.2.1)} \end{cases}$

Table 7-7/G.992.3 – Derived characteristics of the ATU data frame

Symbols	Definition and value
PER_p	**Definition**: The period of the overhead channel in latency path #*p* $$PER_p = \frac{T_p \times S_p \times SEQ_p}{4 \times M_p} ms$$
PMS-TC	**Definition**: Impulse Noise Protection INP_p in number of DMT symbols of latency path function #*p* $$INP_p = \left(\frac{1}{2}\right) \times (S \times D) \times \left(\frac{R}{N\_FEC}\right)$$

7.6.2 Valid framing configurations

Table 7-8 displays the allowable range of each PMS-TC control parameter. Additionally, the control parameters shall satisfy some relationships to one another for the set of control parameter values to be valid as displayed in Table 7-8. Some ranges of the valid control parameter values are expressed in terms of NSC, which is the number of subcarriers as defined in clause 8.

An additional requirement is made on the value of the $B_{p.n}$. Each frame bearer shall be transported in one and only one latency path. This means that in any valid framing configuration, there shall be no more than one non-zero control parameter in each set $\{B_{0.n}, B_{1.n}, B_{2.n}, B_{3.n}\}$.

Table 7-8/G.992.3 – Valid framing configurations

Parameter	Capability
MSG_{min}	$4000 \le MSG_{min} \le 64000$
MSG_{max}	$MSG_{max} = 64000$
N_{BC}	$1 \le N_{BC} \le 4$
N_{LP}	$1 \le N_{LP} \le 4$
MSG_{LP}	$0 \le MSG_{LP} \le 3$
MSG_C	The valid values of MSG_C are those required to support valid minimum and maximum overhead rates, MSG_{min} and MSG_{max}.
$B_{p.n}$	$0 \le B_{p.n} \le 254$, $\sum_n B_{p.n} \le 254$
M_p	1, 2, 4, 8 or 16. If $R_p = 0$ then $M_p = 1$
T_p	$1 \le T_p \le 64$
R_p	0, 2, 4, 6, 8, 10, 12, 14, or 16
D_p	1, 2, 4, 8, 16, 32, 64. If $R_p = 0$ then $D_p = 1$
L_p	$1 \le L_p \le 15 \times (NSC - 1)$ and $\sum L_p$ shall be such that $8 \le \sum L_p \le 15 \times (NSC - 1)$
Relation of S_p and M_p	Configurations that satisfy the following relationship are valid: $M_p / 2 \le S_p \le 32 \times M_p$ (see Note 1).

Table 7-8/G.992.3 – Valid framing configurations

Parameter	Capability
Overhead Rate Constraints	Configurations that satisfy the following relationship are valid: 0.8 kbit/s $\leq OR_p \leq 64$ kbit/s (see Note 2).
Delay Constraints	Configurations that satisfy the following relationship are valid: $\frac{1}{2} \leq S_p \leq 64$ (see Note 3).
Overhead Channel Period	Configurations that provide a period for each overhead channel PER_p between 15 and 20 ms are valid.
NOTE 1 – This condition is a bound on the number of Mux Data Frames per symbol. NOTE 2 – The 0.8 kbit/s overhead rate lowerbound corresponds to an $SEQ_p = 2$ (see Table 7-14) and an overhead channel period of 20 ms. NOTE 3 – This condition puts bounds on the number of FEC codewords per symbol.	

7.6.3 Mandatory configurations

7.6.3.1 Mandatory latency path function

An ATU shall support all combinations of the values of PMS-TC control parameters for latency path function #0 displayed in Tables 7-9 and 7-10 in the downstream and upstream direction, respectively. Configurations that result in non-integer values S_0 shall be supported. The values shown in the tables shall be supported in all transmitter and receivers.

Table 7-9/G.992.3 – Mandatory downstream control parameter support for latency path #0

Parameter	Capability
MSG_{min}	All valid values of MSG_{min} shall be supported within latency path #0.
MSG_{max}	MSG_{max} shall be set to 64000 within latency path #0.
Number of frame bearers	N_{BC}
B_{00}	All valid values of B_{00} shall be supported up to a maximum required to support the highest mandatory downstream data rate for any TPS-TC supported by the ATU.
MSG_{LP}	0
MSG_C	All valid values of MSG_C shall be supported within path #0.
M_0	All valid values of M_0 shall be supported.
T_0	All valid values of T_0 shall be supported.
R_0	All valid values of R_0 shall be supported.
D_0	All valid values of D_0 shall be supported.
L_0	All valid values of L_0 shall be supported up to a maximum required to support the highest mandatory downstream data rate for any TPS-TC supported by the ATU.

Table 7-10/G.992.3 – Mandatory upstream control parameter support for latency path #0

Parameter	Capability
MSG_{min}	All valid values of MSG_{min} shall be supported within latency path #0.
MSG_{max}	MSG_{max} shall be set to 64000 within latency path #0.
Number of frame bearers	N_{BC}
B_{00}	All valid values of B_{00} shall be supported up to a maximum required to support the highest mandatory upstream data rate for any TPS-TC supported by the ATU.
MSG_{LP}	0
MSG_C	All valid values of MSG_C shall be supported within latency path #0.
M_0	All valid values of M_0 shall be supported.
T_0	All valid values of T_0 shall be supported.
R_0	All valid values of R_0 shall be supported.
D_0	All valid values of D_0 shall be supported such that $Dp \leq 8$.
L_0	All valid values of L_0 shall be supported up to a maximum required to support the highest mandatory upstream data rate for any TPS-TC supported by the ATU.

7.6.3.2 Other latency path functions

An ATU shall support all combinations of the values of PMS-TC control parameters for each optional latency path #*p* that is supported as displayed in Tables 7-11 and 7-12 in the downstream and upstream direction, respectively. Configurations that result in non-integer values S_P shall be supported. The values shown in the tables shall be supported in transmitter and receiver.

Table 7-11/G.992.3 – Mandatory downstream control parameter support for optional latency paths

Parameter	Capability
MSG_{min}	All valid values of MSG_{min} shall be supported within any supported latency path.
MSG_{max}	MSG_{max} shall be set to 64000 within any supported latency path.
Number of frame bearers	N_{BC}
B_{p0}	All valid values of B_{p0} shall be supported up to a maximum required to support the highest mandatory downstream data rate for any TPS-TC supported by the ATU.
MSG_{LP}	Any supported latency path function shall be capable of carrying the message based portion of the overhead structure. $MSG_{LP} = p$ shall be supported.
MSG_C	All valid values of MSG_C shall be supported within any supported latency path.
M_P	All valid values of M_p shall be supported.
T_p	All valid values of T_p shall be supported
R_p	$R_{p\ max}$ is identified during initialization. All valid values of R_p up to and including $R_{p\ max}$ shall be supported.
D_p	$D_{p\ max}$ is identified during initialization. All valid values of D_p up to and including $D_{p\ max}$ shall be supported.
L_p	All valid values of L_p shall be supported up to a maximum required to support the highest mandatory downstream data rate for any TPS-TC supported by the ATU.

Table 7-12/G.992.3 – Mandatory upstream control parameter support for optional latency paths

Parameter	Capability
MSG_{min}	All valid values of MSG_{min} shall be supported within any supported latency path.
MSG_{max}	MSG_{max} shall be set to 64000 within any supported latency path.
Number of frame bearers	N_{BC}
B_{p0}	All valid values of B_{p0} shall be supported up to a maximum required to support the highest mandatory upstream data rate for any TPS-TC supported by the ATU.
MSG_{LP}	Any supported latency path function shall be capable of carrying the message based portion of the overhead structure. $MSG_{LP} = p$ shall be supported.
MSG_C	All valid values of MSG_C shall be supported within any supported latency path.
M_p	All valid values of M_p shall be supported.
T_p	All valid values of T_p shall be supported
R_p	$R_{p\ max}$ is identified during initialization. All valid values of R_p up to and including $R_{p\ max}$ shall be supported.
D_p	$D_{p\ max}$ is identified during initialization. All valid values of D_p up to and including $D_{p\ max}$ shall be supported.
L_p	All valid values of L_p shall be supported up to a maximum required to support the highest mandatory upstream data rate for any TPS-TC supported by the ATU.

7.7 Data plane procedures

7.7.1 Latency path function

7.7.1.1 Mux data frame selector

Within latency path function #p, the Mux Data Frame Selector multiplexes the frame bearers with the overhead channel for latency path function #p. The output of the Mux Data Frame Selector is in the structure of the Mux Data Frame at Reference Point A. The control parameters M_p, T_p, and $B_{p0}, \ldots, B_{p3}$ determine the selection and the order of the octets from Frame.Bearer(n).confirm primitives, the CRC octet described in 7.7.1.2, and the Overhead Channel #p from the Overhead Access Function described in 7.8.2.

The Mux Data Frame Selector maintains a counter that is initialized to zero at the completion of initialization. The counter is incremented each time a complete Mux Data Frame is constructed and is used in conjunction with the control parameter T_p in the following manner. The first octet of every Mux Data Frame is nominally used to transport the shared overhead channel of the PMS-TC function. However, this octet is used to carry data sometimes if the value of T_p is not 1. If T_p is not one and if the counter value modulo T_p is zero, then the octet is used to transport overhead; otherwise an extra octet of data is transported. The data is taken from the frame bearer with the lowest index that is assigned to latency path #p. In the case that there is no frame bearer assigned to latency path #p, an octet with the value of zero is used.

When the octet is used for overhead, the next octet is taken from the overhead message structure described in 7.8.2.1. Because the counter used in conjunction with T_p is reset at the completion of initialization, the first Mux Data Frame generated always has a sync octet carrying the overhead channel.

The remaining octets of every Mux Data Frame in latency path #p are constructed by taking B_{p0} octets from Frame.Bearer(0).confirm primitives, B_{p1} octets from Frame.Bearer(1), etc. The octets are taken from the primitives so that their octet alignment, MSB position, and order within the frame bearer are maintained. Each Mux Data Frame always contains a total of K_p octets.

The Mux Data Frame Selector procedure of the latency path function #p creates M_p Mux Data Frames, a total of $M_p \times K_p$ octets. This procedure is followed by the CRC procedure.

7.7.1.2 Cyclic redundant checksum

Each latency path periodically calculates a CRC octet, *crc*0 to *crc*7, to enable error detection. The CRC covers $T_P \times SEQ_p \times K_p - 1$ message octets, starting from the first octet after the sync octet of the first Mux Data Frame and ending with the last octet of the last Mux Data Frame.

The *crc*0 to *crc*7 bits shall be computed from $(T_P \times SEQ_p \times K_p - 1) \times 8$ message bits at Reference Point A using the equation:

$$crc(D) = M(D)D^8 \text{ modulo } G(D)$$

where:

$$M(D) = m_0 Dk^{-1} + m_1 D^{k-2} + ... + m_{k-2} D + m_{k-1}, \text{ is the message polynomial,}$$

$$k = (T_p \times SEQ_p \times K_p - 1) \times 8,$$

$$G(D) = D^8 + D^4 + D^3 + D^2 + 1, \text{ is the generating polynomial,}$$

$$crc(D) = c_0 D^7 + c_1 D^6 + ... + c_6 D + c_7, \text{ is the check polynomial,}$$

and D is the delay operator.

That is, the CRC is the remainder when $M(D)\ D^8$ is divided by $G(D)$. Each octet shall be input into the $crc(D)$ equation least significant bit first.

The CRC value calculated is presented to the Mux Data Frame Selector described in 7.7.1.1 for transport during the next available overhead channel octet, i.e., first octet in the next repetition of the overhead channel structure (see 7.8.2.1). This procedure is followed by the scrambler procedure.

7.7.1.3 Scrambler

The binary data streams at Reference Point A shall be scrambled as illustrated in Figure 7-9 using the following equation:

$$d_n' = d_n \oplus d_{n-18}' \oplus d_{n-23}'$$

where d_n is the n-th input to the scrambler,

and d_n' is the n-th output from the scrambler.

Each octet shall be input into the scrambler equation least significant bit first. The scrambler procedure of the latency path function #p shall scramble M_p Mux Data Frames, or $M_p \times K_p$ octets. This procedure is followed by the FEC procedure.

NOTE – The starting state of the scrambler is not specified. Receiver implementations should use self-synchronizing descrambler designs.

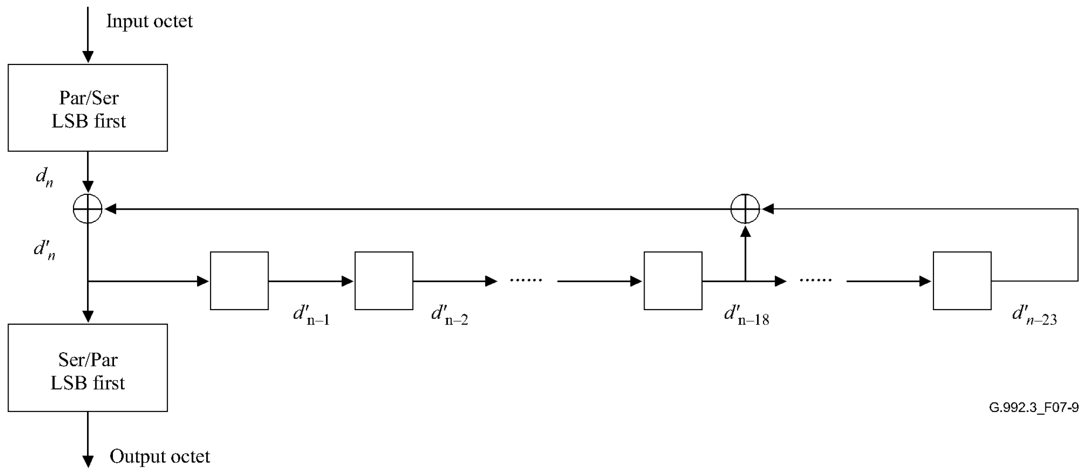

Figure 7-9/G.992.3 – Scrambler procedure

7.7.1.4 Forward error correction function

The FEC procedure inserts Reed-Solomon FEC redundancy octets to provide coding gain as an outer coding function to the PMD function. The FEC procedure of latency path function #*p* shall calculate R_p octets from $M_p \times K_p$ input octets. The octets are appended to the end of the input octets in the structure of FEC Output Data Frame at Reference Point B.

When $R_p = 0$, no redundancy octets are appended and the values in the FEC Output Data Frame are identical to the input values. For all other values of R_p, the following encoding procedure shall be used to create the R_p octets:

The FEC procedure shall take in M_p scrambled Mux Data Frames comprising message octets, m_0, m_1, … , $m_{Mp \times Kp-2}$, $m_{Mp \times Kp-1}$. The procedure shall produce R_p redundancy octets c_0, c_1, … , c_{Rp-2}, c_{Rp-1}. These two taken together comprise the FEC codeword of size $M_p \times K_p + R_p$ octets. The R_p redundancy octets shall be appended to the message octets to form the FEC Output Data Frame at Reference Point B.

At the end of the initialization sequence, the FEC Function always starts with the first of M_p Mux Data Frames.

The redundancy octets are computed from the message octets using the equation:

$$C(D) = M(D)D^{Rp} \text{ modulo } G(D)$$

where:

$M(D) = m_0 D^{Mp \times Kp-1} + m_1 D^{Mp \times Kp-2} + \ldots + m_{Mp \times Kp-2} D + m_{Mp \times Kp-1}$ is the message polynomial,

$C(D) = c_0 D^{Rp-1} + c_1 D^{Rp-2} + \ldots + c_{Rp-2} D + c_{Rp-1}$ is the check polynomial, and

$G(D) = \prod \left(D + \alpha^i\right)$ is the generator polynomial of the Reed-Solomon code,

where the index of the product runs from $i = 0$ to $R_p - 1$.

That is, *C(D)* is the remainder obtained from dividing *M*(D) D^R by *G(D)*. The arithmetic is performed in the Galois Field GF(256), where α is a primitive element that satisfies the primitive binary polynomial $x^8 + x^4 + x^3 + x^2 + 1$. A data octet ($d_7$, d_6, … , d_1, d_0) is identified with the Galois Field element $d_7\alpha^7 + d_6\alpha^6 \ldots + d_1\alpha + d_0$.

The FEC procedure of the latency path #p creates $N_{FEC.p}$ octets in the structure of a FEC Output Data Frame at Reference Point B. This procedure is followed by the interleaver procedure.

7.7.1.5 Interleaver

To spread the Reed-Solomon codeword and therefore reduce the probability of failure of the FEC in the presence of impulse noise, the FEC Output Data Frames shall be convolutionally interleaved. The interleaver creates the Interleaved FEC Output Data Frames at Reference point C, at the output of the latency path function. This procedure is followed by the frame multiplexing procedure.

Convolutional interleaving is defined by the rule (using the currently defined values of the framing control parameters D_p and the derived parameter $N_{FEC.p}$):

> Each of the $N_{FEC.p}$ octets B_0, B_1, … , $B_{NFEC.(p-1)}$ in an FEC Output Data Frame is delayed by an amount that varies linearly with the octet index. More precisely, octet B_i (with index i) is delayed by $(D_p - 1) \times i$ octets, where D_p is the interleaver depth.

An example for $N_{FEC.p} = 5$, $D_p = 2$ is shown in Table 7-13, where B_i^j denotes the i-th octet of the j-th FEC Output Data Frame.

Table 7-13/G.992.3 – Convolutional interleaving example for $N_{FEC.p} = 5$, $D_p = 2$

Interleaver input	B_0^j	B_1^j	B_2^j	B_3^j	B_4^j	B_0^{j+1}	B_1^{j+1}	B_2^{j+1}	B_3^{j+1}	B_4^{j+1}
Interleaver output	B_0^j	B_3^{j-1}	B_1^j	B_4^{j-1}	B_2^j	B_0^{j+1}	B_3^j	B_1^{j+1}	B_4^j	B_2^{j+1}

With the above-defined rule, the output octets from the interleaver always occupy distinct time slots when $N_{FEC.p}$ is odd and D_p is a power of 2. When $N_{FEC.p}$ is even, a dummy octet shall be added at the beginning of the codeword at the input to the interleaver. The resultant odd-length codeword is then convolutionally interleaved, and the dummy octet shall then be removed from the output of the interleaver.

The interleaving procedure of the latency path function #p shall interleave a single FEC Output Data Frame, or $M_p \times K_p + R_p$ octets. This procedure is followed by the Frame Multiplexing procedure.

7.7.2 Frame multiplexing

The output signals of all latency paths are multiplexed together to form the output of the PMS-TC function. The frame multiplexing procedure combines bits from each configured latency path in decreasing label order, starting from $p = 3$ down to $p = 0$. L_p bits are taken from each latency path. $L_p = 0$ if latency path #p is not supported or disabled. The bits are taken LSB first. The data is packed into a PMD.Bits.confirm primitive in order of latency path $p = 3$ down to $p = 0$.

7.8 Control plane procedures

7.8.1 NTR transport

An ATU-C may optionally transport an 8 kHz timing marker as NTR to support the transport of a timing reference from voice PSTN access network to equipment located with the ATU-R. The 8 kHz timing marker is provided to the ATU-C as part of the interface at the V reference point. Additionally, if this capability is supported, the local PMD shall provide a PMD sampling clock that is a multiple of 2.208 MHz ±50 ppm along with an indication of when each overhead message structure (described in 7.8.2.1) begins.

If NTR transport is configured during initialization or reconfiguration of the PMS-TC function, the ATU-C shall generate an 8 kHz local timing reference (LTR) by dividing the PMD sampling clock

The receive PMD function should not use an excessive number of monitored subcarriers (i.e., subcarriers in the MEDLEYset to which it allocates $b_i = 0$ and $g_i > 0$) to aid in the conservation of spectrum.

These requirements on the bits and gains table apply in the L0 state and at entry into the L2 state. The L2 entry grant response message indicates the gains table to be used in the L2 state (see 9.4.1.7). However, at entry into the L2 state, the excess margin may not be minimized. Power trimming during the L2 state may be used to minimize the excess margin. The L2 entry and trim grant response messages indicate the PCB value to be used in the L2 state (see 9.4.1.7). Power trimming is defined as changing the downstream power cutback (*PCBds*) level, resulting in a change of the downstream reference transmit PSD (*REFPSDds*) level. Power trimming changes the *PCBds* value used during the L2 state and does not change the gi values determined at the time of entry into the L2 state.

The g_i values in dB shall be defined as the 20 log g_i (g_i in linear scale). A g_i value of –14.5 dB corresponds to a g_i of 0.1888 in linear scale. A g_i value of +2.5 dB corresponds to a g_i value of 1.333 in linear scale. Same relationship shall be used for the tss_i values in dB and in linear scale.

NOTE – The g_i define a scaling of the root mean square (rms) subcarrier power levels relative to the *REFPSD* level (see 8.13.5). They are independent of any methods that manufacturers may use to simplify implementation (e.g., constellation nesting).

8.7 Constellation encoder for synchronization and L2 exit symbols

The constellation encoder for the synchronization and L2 exit symbols is shown as part of the transmit PMD function in Figure 8-5. A synchronization or L2 exit symbol shall either be an SS-REVERB symbol or an SS-SEGUE symbol.

Clauses 8.7.1 and 8.7.2 shall define respectively the constellation mapper and gain scaling for an SS-REVERB symbol. An SS-SEGUE symbol shall be defined as a subcarrier-by-subcarrier 180 degrees phase reversal of an SS-REVERB symbol (i.e., an SS-SEGUE symbol modulates the bitwise inverted REVERB PRBS data pattern).

The transmit PMD function transports the following types of PMD.Synchflag.request primitives (as received from the transmit PMS-TC function) for synchronization of:

- On-line reconfiguration during the L0 state (see 8.7.3);
- Entry from the L0 into the L2 power management state (see 8.7.4);
- Exit from the L2 power management into the L0 state (see 8.7.6);
- Power trimming during the L2 state (see 8.7.5).

8.7.1 Constellation mapper

For the subcarriers in the MEDLEYset, the REVERB PRBS data pattern shall be mapped on the SS-REVERB symbols in the same way as it is mapped on the REVERB symbols during the REVERB1 state (see 8.13.4.1.1). Two bits are mapped on each of the subcarriers, generating a 4-QAM constellation point for each of the subcarriers, i.e., X_i and Y_i for index $i = 1$ to $NSC - 1$.

The values of X and Y of the 4-QAM constellation points as shown in the constellation diagrams are on a ± 1 grid. These values require appropriate scaling such that, at the output of the constellation mapper, all constellations represent the same rms energy as a subcarrier transmitted at the reference transmit PSD level (*REFPSD*).

For the subcarriers not in the MEDLEYset, the constellation mapper may select a discretionary (X, Y) point (which may change from symbol to symbol and which does not necessarily coincide with a constellation point).

8.7.2 Gain scaling

In the L0 state, gain scaling shall be applied to synchronization symbols in the same way as it is applied to data symbols in the L0 state (see 8.6.4).

In the L2 state, gain scaling shall be applied to synchronization symbols in the same way as it is applied to data symbols in the L2 state (see 8.6.4).

In the L2 state, gain scaling shall be applied to L2 exit symbols, as indicated in the L2 entry or L2 trim grant response message related to the last previously transmitted PMD.Synchflag primitive (see 9.4.1.7). The L2 entry grant response message indicates whether the L0 or L2 state gain scaling table is to be used with the L2 exit symbols. The L2 entry and L2 trim grant response messages indicate the *PCBds* value to be used with the L2 exit symbols (see 9.4.1.7).

8.7.3 On-line reconfiguration during the L0 state

The PMD transmit function inserts a synchronization symbol every 68 data symbols, as defined in 8.4. The synchronization symbols shall be transmitted at symbolcount 68, and:

- permit the PMD receive function to recover the PMS-TC frame boundary after micro-interruptions that might otherwise force re-initialization;
- provide a time marker for the on-line reconfiguration during the L0 state.

Every time the transmit PMD function receives a PMD.Synchflag.request primitive (related to on-line reconfiguration during the L0 state) from the transmit PMS-TC layer, the phase of the first next inserted synchronization symbol shall be inverted, and remain inverted until the next PMD.Synchflag.request primitive is to be carried. At the start of Showtime, the first synchronization symbol transmitted shall be an SS-REVERB symbol.

8.7.4 Entry from the L0 into the L2 power management state

Every time the transmit PMD function receives a PMD.Synchflag.request primitive (related to entry from the L0 into the L2 power management state) from the transmit PMS-TC layer, the phase of the first next inserted synchronization symbol shall be inverted, and remain inverted until the next PMD.Synchflag.request primitive is to be carried.

Prior to entry from the L0 into the L2 power management state, the ATU shall store the downstream control parameters which need to be restored at exit from the L2 into the L0 power management state.

The receive PMD function can distinguish PMD.Synchflag primitives related to entry from the L0 into the L2 power management from those related to on-line reconfiguration and those related to L2 power trimming based on previously exchanged information between the management entities.

8.7.5 Power trimming during the L2 state

Every time the transmit PMD function receives a PMD.Synchflag.request primitive (related to power trimming during the L2 state) from the transmit PMS-TC layer, the phase of the first next inserted synchronization symbol shall be inverted, and remain inverted until the next PMD.Synchflag.request primitive is to be carried.

The receive PMD function can distinguish PMD.Synchflag primitives related to L2 power trimming from those related to L0 on-line reconfiguration and those related to entry from the L0 into the L2 power management based on previously exchanged information between the management entities.

International Telecommunication Union

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TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.993.1

(06/2004)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

Digital sections and digital line system – Access networks

Very high speed digital subscriber line transceivers

ITU-T Recommendation G.993.1

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ITU-T G-SERIES RECOMMENDATIONS

TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation G.993.1

Very high speed digital subscriber line transceivers

Summary

G.993.1 VDSL (Very high speed Digital Subscriber Line) permits the transmission of asymmetric and symmetric aggregate data rates up to tens of Mbit/s on twisted pairs. G.993.1 includes worldwide frequency plans that allow asymmetric and symmetric services in the same group of twisted pairs (known as a binder). G.993.1 transceivers must overcome many types of ingress interference from radio and other transmission techniques that occur in the same frequencies of typical deployment scenarios. Similarly, G.993.1 transmission power transmission levels have been designed to minimize potential egress interference into other transmission systems. As with other Recommendations in the G.99x series, G.993.1 uses G.994.1 to handshake and initiate the transceiver training sequence.

Source

ITU-T Recommendation G.993.1 was approved on 13 June 2004 by ITU-T Study Group 15 (2001-2004) under the ITU-T Recommendation A.8 procedure.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g. interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

7.2.1 Data flow

The eoc data flow includes two contra-directional streams of 2-octet blocks each (eoc_tx, eoc_rx) with independent rates flowing between the eoc application layer (VME) and TPS-TC OC block (OC-TC). The bit rates of both streams shall not exceed the predefined upper limit of the OC channel aggregate transport capability. The data flow signal description is presented in Table 7-1.

If data streams are *serial* by implementation, the MSB of each octet shall be sent first.

7.2.2 Synchronization flow

This flow provides synchronization between the eoc application layer (VME) and the OC-TC (see 10.3.1). The flow includes the following synchronization signals, presented in Table 7-2:

- transmit and receive timing signals (eoc_tx_clk, eoc_rx_clk): both asserted by the eoc processor;
- transmit enable flag (tx_enbl): asserted by OC-TC and allows to transmit the next 2-octet block;
- receive enable flag (rx_enbl): asserted by OC-TC and indicates that the next 2-octet block is allocated in the OC-TC receive buffer.

Table 7-2/G.993.1 – OC-TC: γ interface data and synchronization flow summary

Signal	Description	Direction	Notes
		Data flow	
eoc_tx	Transmit eoc data	VME → OC-TC	Two-octet block
eoc_rx	Receive eoc data	VME ← OC-TC	
		Synchronization flow	
eoc_tx_clk	Transmit clock	VME → OC-TC	
eoc_rx_clk	Receive clock	VME → OC-TC	
tx_enbl	Transmit enable flag	VME ← OC-TC	
rx_enbl	Receive enable flag	VME ← OC-TC	

NOTE – The main buffering required to implement the eoc communication protocol should be provided by the VME; only a minimum buffering for eoc is supposed in OC-TC.

8 PMS-TC sublayer

The PMS-TC sublayer provides transmission medium specific TC functions, such as framing, Forward Error Correction (FEC), and interleaving.

8.1 PMS-TC functional model

All data bytes shall be transmitted MSB first. All serial processing (e.g., scrambling, CRC calculation) shall however be performed LSB first, with the outside world MSB considered as the VDSL LSB. As a result, the first incoming bit (outside world MSB) shall be the first bit processed inside VDSL (VDSL LSB). The PMS-TC functional diagram is presented in Figure 8-1.

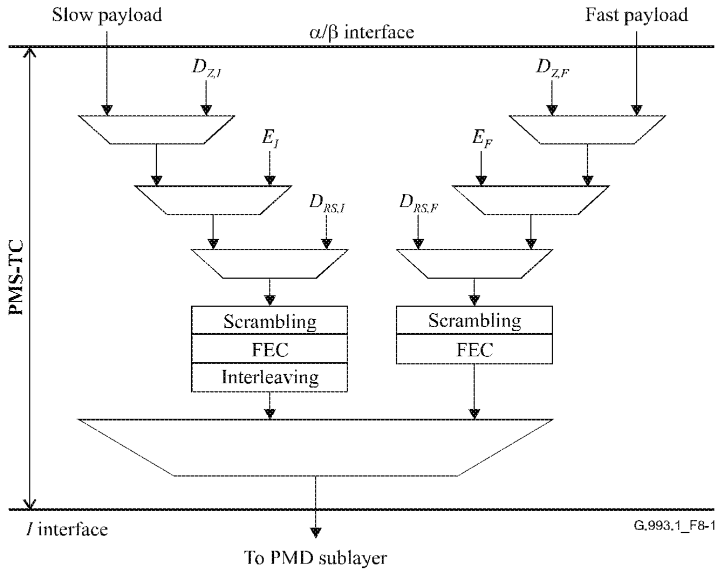

Figure 8-1/G.993.1 – Diagram of PMS-TC sublayer

8.2 Scrambler

A scrambler shall be used to reduce the likelihood that a long sequence of zeros will be transmitted over the channel. The scrambler shall be self-synchronizing such that descrambling can occur without requiring a particular alignment with the scrambled sequence. The scrambler shall be represented by the equation below, where $m(n)$ is a message bit sample at sample time n and the output of the scrambler $x(n)$ shall be given by:

$$x(n) = m(n) + x(n-18) + x(n-23)$$

All arithmetic shall be modulo 2. As long as the scrambler is initialized with values other than zero, an "all zeros" sequence for $m(n)$ will result in a pseudo-random sequence of length $2^{23} - 1$.

8.3 Forward error correction

A standard byte-oriented Reed-Solomon code shall be used to provide protection against random and burst errors.

A Reed-Solomon code word contains $N = K + R$ bytes, comprised of R redundant check bytes c_0, c_1, ..., c_{R-2}, c_{R-1} appended to K message bytes m_0, m_1, ..., m_{K-2}, m_{K-1}. The check bytes shall be computed from the message bytes using the equation

$$C(D) = M(D)D^R \bmod G(D)$$

where:

$M(D) = m_0 D^{K-1} \oplus m_1 D^{K-2} \oplus \ldots \oplus m_{K-2} D \oplus m_{K-1}$ is the message polynomial

$C(D) = c_0 D^{R-1} \oplus c_1 D^{R-2} \oplus \ldots \oplus c_{R-2} D \oplus c_{R-1}$ is the check polynomial

$G(D) = \prod (D \oplus \alpha^i)$ is the generator polynomial of the Reed-Solomon code, where the index of the product runs from i = 0 to $R - 1$.

This means that $C(D)$ is the remainder obtained from dividing $M(D)D^R$ by $G(D)$. The arithmetic shall be performed in the Galois Field GF(256), where α is a primitive element that satisfies the

primitive binary polynomial $x^8 \oplus x^4 \oplus x^3 \oplus x^2 \oplus 1$. A data byte $(d_7, d_6, \ldots, d_1, d_0)$ is identified with the Galois Field element $d_7\alpha^7 \oplus d_6\alpha^6 \oplus \ldots \oplus d_1\alpha \oplus d_0$.

Both K and R shall be programmable parameters. Redundancy values of R = 0, 2, 4, 6, 8 … 16 shall be supported. The following codeword parameters specified as (N,K) shall be supported: (144,128) and (240,224). Other values for N and K are optional. However, N shall be less than or equal to 255.

8.4 Interleaving

8.4.1 General

Interleaving shall be used to protect the data against bursts of errors by spreading the errors over a number of Reed-Solomon codewords. The interleave depth shall be programmable with a maximum interleave depth of 64 codewords when the number of octets per codeword (N) equals 255. For smaller values of N the interleave depth can grow nearly proportionately.

It shall be possible to adjust the interleave depth via the management system to meet latency requirements. The latency of the slow path is a function of the data rate and burst error correction capability. For data rates greater than or equal to 13 Mbit/s, the latency between the α and β interfaces shall not exceed 10 ms when the interleaver depth is set to the maximum. At lower data rates there is a trade-off between higher latency and decreased burst error correction ability. At any data rate, the minimum latency occurs when the interleaver is turned off.

When the interleaver is on, the codewords shall be interleaved before transmission to increase the immunity of RS codewords to bursts of errors. The convolutional interleaver is defined by two parameters: the interleaver block length, I, and the interleaving depth, D. The block length I shall divide the RS codeword length N (i.e., N shall be an integer multiple of I). The convolutional interleaver uses a memory in which a block of I octets is written while an (interleaved) block of I octets is read. Details of the implementation are given in 8.4.2.

The same size interleaving memory (see Table 8-1) is needed for interleaving at the transmitter and de-interleaving at the receiver.

The convolutional interleaving introduces an absolute read-to-write delay, Δ_J, that increments linearly with the octet index within a block of I octets:

$$\Delta_j = (D-1) \times j$$

where $j = 0, 1, 2, \ldots, I-1$.

8.4.2 Triangular implementation

To decrease the implementation complexity, the delay increment $(D-1)$ shall be chosen as a multiple of the interleaver block length (I), i.e.: $D-1 = M \times I$. The $(D-1)$ to I ratio is the interleaving depth parameter (M). The characteristics of convolutional interleaving are shown in Table 8-1. The parameters t and q depend on the characteristics of the RS code and are defined as:

– t = number of bytes that can be corrected by RS codewords = half the number of redundancy bytes = $R/2$;

– q = length of RS codeword divided by the length of an interleaver block = N/I.

Table 8-1/G.993.1 – Characteristics of convolutional interleaving

Parameter	Value
Interleaver block length (I)	I bytes (equal to or divisor of N)
Interleaving Depth (D)	$M \times I + 1$
(De)interleaver memory size	$M \times I \times (I - 1)/2$ bytes
Correction capability	$\lfloor t/q \rfloor \times (M \times I + 1)$ bytes
End-to-end delay	$M \times I \times (I - 1)$ bytes

The example in Figure 8-2 shows $I = 7$. I parallel branches (numbered $0 \ldots I - 1$) are implemented with a delay increment of M octets per branch. Each branch shall be a FIFO shift register (delay line) with length $0 \times M \ldots (I - 1) \times M$ bytes. The deinterleaver is similar to the interleaver, but the branch indices are reversed so that the largest interleaver delay corresponds to the smallest deinterleaver delay. Deinterleaver synchronization shall be achieved by routing the first byte of an interleaved block of I bytes into branch 0.

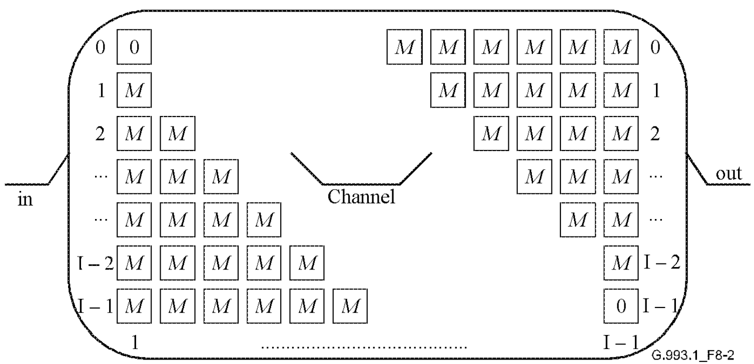

Figure 8-2/G.993.1 – Implementation example with $D - 1 = M \times I$ and $I = 7$

Table 8-2/G.993.1 – Example of interleaver parameters with RS(144,128)

Rate [kbit/s]	Interleaver parameters	Interleaver depth	(De)interleaver memory size	Erasure correction	End-to-end delay
50 × 1024	$I = 72$ $M = 13$	937 blocks of 72 bytes	33 228 bytes	3 748 bytes 520 µs	9.23 ms
24 × 1024	$I = 36$ $M = 24$	865 blocks of 36 bytes	15 120 bytes	1 730 bytes 500 µs	8.75 ms
12 × 1024	$I = 36$ $M = 12$	433 blocks of 36 bytes	7 560 bytes	866 bytes 501 µs	8.75 ms
6 × 1024	$I = 18$ $M = 24$	433 blocks of 18 bytes	3 672 bytes	433 bytes 501 µs	8.5 ms
4 × 1024	$I = 18$ $M = 16$	289 blocks of 18 bytes	2 448 bytes	289 bytes 501 µs	8.5 ms
2 × 1024	$I = 18$ $M = 8$	145 blocks of 18 bytes	1 224 bytes	145 bytes 503 µs	8.5 ms

The following interleaver parameters shall be supported:

– For $(N,K) = (144,128)$ the following values for M and I shall be supported:
  $I = 36$ and M between 2 and 52.
– For $(N,K) = (240,224)$ the following values for M and I shall be supported:
  $I = 30$ and M between 2 and 62.

8.5 Framing

8.5.1 Frame description

A *frame* is a set of bytes carried by one DMT symbol. The frame frequency depends on the total length of the cyclic extension (see 9.2.2). A frame shall be composed of two sources: the "fast" buffer and the "interleaved" (or "slow") buffer. The index i refers to parameters related to the fast or interleaved buffers ($i \in \{F, I\}$). The inclusion of the fast buffer shall be optional. When the fast buffer is not included, the interleaved buffer shall have the capability to carry non-interleaved data by setting the interleaver depth to zero.

Both fast and interleaved buffer shall contain an integer number of RS-encoded bytes. Neither the fast nor the interleaved buffer is required to carry an integer number of RS codewords. To reduce the end-to-end delay, it is recommended that the fast buffer (or the interleaved buffer when the interleaver depth is zero) carries at least one RS codeword. The framing parameters shall be exchanged between the VTU-O and VTU-R during initialization.

The framing rules described in this clause are represented in Figure 8-3.

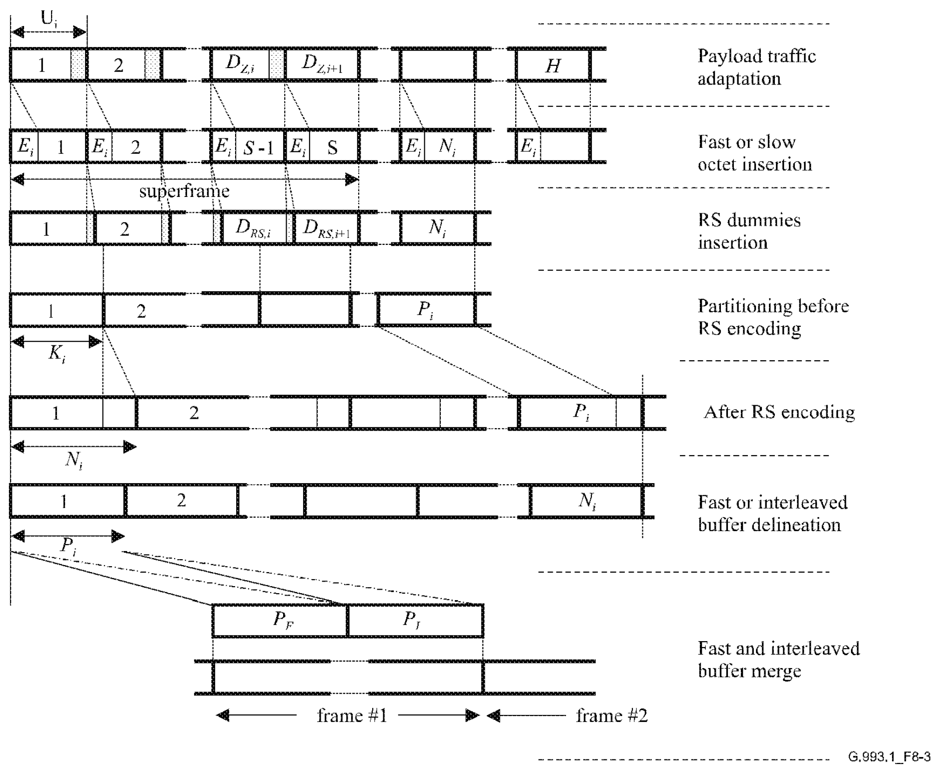

Figure 8-3/G.993.1 – Framing description

8.5.2 Payload adaptation

The α/β interface provides bytes at a rate multiple of 64 kbit/s. In order to map an integer number of bytes into a frame, the TPS-TC byte flow shall be stuffed with the appropriate number of dummies.

For an $n_i \times 64$ kbit/s rate, we have on average $n_i \times 8000/f_s$ bytes per frame, with f_s the symbol frequency. This number will not be integer for a general value of f_s. Since the cyclic extension $L_{CP} + L_{CS} - \beta$ is a multiple of 2^{n+1} however, (see 9.2.2), we always have an integer number of bytes every $H = 138$ frames. If we define k as:

$$k = \frac{8\ kbytes \times H}{f_s}$$

we can transport $n_i \times k$ TPS-TC payload bytes in H frames. In order to transport an integer number of bytes per frame, we have to insert an appropriate number of dummy bytes. Every frame will contain a total of U_i bytes (TPS-TC bytes + dummy bytes), with:

$$U_i = \left\lceil \frac{n_i \times k}{H} \right\rceil$$

The number of dummy bytes $D_{Z,i}$ to be inserted every H packets shall therefore be:

$$D_{Z,i} = \left\lceil \frac{n_i \times k}{H} \right\rceil \times H - (n_i \times k)$$

These dummy bytes shall be inserted in the last position of the first $D_{Z,i}$ packets of U_i bytes in a sequence of H packets. The value of the $D_{Z,i}$ dummies shall be 0x3A.

8.5.3 RS encoding

After payload adaptation, E_i overhead bytes (see 8.5.5) shall be added to the head-end of each packet of U_i bytes (see Figure 8-3). These bytes are called fast and slow bytes for the fast and slow channel respectively. Next, a sequence of N_i packets of $(E_i + U_i)$ bytes shall be RS-encoded. In order to achieve an integer number of RS-codewords per N_i packets, RS-dummy bytes may have to be inserted. The RS-codeword length is equal to the parameter N_i.

The number of RS-encoded bytes, B_i, per N_i packets is given by:

$$B_i = \left[N_i \times (E_i + U_i) + D_{RS,i}\right] \times \frac{N_i}{K_i}$$

In the above equation, the parameter N_i denotes both the number of packets of $(E_i + U_i)$ bytes and also the length of a RS-codeword (in bytes). The parameter K_i is the number of information bytes in an RS-codeword.

The number of RS dummy bytes, $D_{RS,i}$, inserted to carry an integer number of RS-codewords in every N_i frames is given by

$$D_{RS,i} = \left\lceil \frac{N_i \times (E_i + U_i)}{K_i} \right\rceil \times K_i - N_i \times (E_i + U_i)$$

Each one of the $D_{RS,i}$ dummies shall be inserted at the tail-end of the first $D_{RS,i}$ packets of $(E_i + U_i)$ bytes in a sequence of N_i packets (see Figure 8-3). The value of the $D_{RS,i}$ bytes shall be 0xD3.

After RS-dummy insertion, the number of RS-encoded bytes per frame carried in either the fast or interleaved buffer is given by:

$$P_i = \frac{B_i}{N_i} = \frac{N_i \times (E_i + U_i) + D_{RS,i}}{K_i} = \left\lceil \frac{N_i \times (E_i + U_i)}{K_i} \right\rceil$$

NOTE – The parameter $B_i = P_i N_i$ represents both the number of bytes in N_i frames (with P_i bytes per frame) and also the number of bytes in P_i codewords (with N_i bytes per codeword). See Figure 8-3.

8.5.4 Definition of superframe

A superframe shall be composed of 10 packets of $U_i + E_i$ bytes.

8.5.5 Contents of fast and slow bytes

Each of the packets in a superframe shall transport E_i overhead bytes, called fast or slow bytes, depending on the channel. The content of these bytes is summarized in Table 8-3. If the fast buffer is empty, the F-EOC bytes shall be transported in the S-EOC bytes. Otherwise, the S-EOC bytes shall be replaced with payload bytes.

There shall be V VOC bytes per packet. They shall always be transported in the slow channel. A setting of $V = 1$ shall be supported, other values for V should be allowed as optional. The value of V shall be exchanged during initialization (see 12.4.6.2.1.1).

If the fast path is active, the NTR byte in the slow channel shall be replaced with a dummy byte. Similarly for the IB bytes.

The fast and slow dummy bytes shall have the value 0xFF.

Table 8-3/G.993.1 – Contents of fast and slow bytes

Packet	Fast bytes		Slow bytes		
	First byte	Other bytes (if any)	First byte	2nd up to (V + 1)st byte	Other bytes (if any)
1	F-CRC	F-EOC	S-CRC	VOC	S-EOC/payload
2	Synch byte	F-EOC	Synch byte	VOC	S-EOC/payload
3-5	IB	F-EOC	IB/dummy	VOC	S-EOC/payload
6	NTR	F-EOC	NTR/dummy	VOC	S-EOC/payload
7-10	Dummy	F-EOC	Dummy	VOC	S-EOC/payload

8.5.5.1 Cyclic Redundancy Check (CRC)

Two cyclic redundancy checks (CRC) – one for the fast buffer and one for the interleaved buffer – shall be generated for each superframe and shall be transmitted in the first packet of the following superframe (see Table 8-3). The CRC byte for the first superframe shall be set to zero.

Eight bits per buffer type (fast or interleaved) and per superframe shall be allocated to the CRC check bits. These bits shall be computed from the k message bits using the equation:

$$crc(D) = M(D)\, D^8 \text{ modulo } G(D)$$

where:

$M(D) = m_0 D^{k-1} + m_1 D^{k-2} + \ldots + m_{k-2} D + m_{k-1}$ is the message polynomial

$G(D) = D^8 + D^4 + D^3 + D^2 + 1$ is the generating polynomial

$crc(D) = c_0 D^7 + c_1 D^6 + \ldots + c_6 D + c_7$ is the check polynomial

D is the delay *operator*.

That is, crc(D) shall be the remainder when $M(D)D^8$ is divided by $G(D)$.

The bits covered by the crc shall include:

– fast buffer: all bits of the fast buffer before RS encoding, except the crc;

– interleaved buffer: all bits of the interleaved buffer before RS encoding, except the crc.

Each byte shall be clocked into the CRC least significant bit first.

8.5.5.2 Synchronization byte

The synchronization byte has the value 0x3C. This synchronization byte shall be used to monitor the frame synchronization.

8.5.5.3 Indicator Bits (IB)

The indicator bits are used to transmit far-end defects and anomalies. The description of the content of the three indicator bytes shall be as summarized in Table 8-4. If the fast channel is active, the indicator bytes shall be transmitted in this channel and the indicator bytes in the slow channel shall be replaced by dummies (having value 0xFF, see 8.5.5).

Table 8-4/G.993.1 – Content of indicator bits

Byte #	Bit #	Definition
1	b0-b7	Reserved for future use
2	b0	Febe-s
	b1	Ffec-s
	b2	Febe-f
	b3	Ffec-f
	b4	Flos
	b5	Rdi
	b6	Fpo
	b7	Flpr
3	b0	LoM (Loss of Margin)
	b1	Fhec-s (used for ATM only, shall be set to 0 for PTM)
	b2	Fhec-f (used for ATM only, shall be set to 0 for PTM)
	b3	Fncd-s/Focd-s (used for ATM only, shall be set to 0 for PTM)
	b4	Fncd-f/Focd-f (used for ATM only, shall be set to 0 for PTM)
	b5-b7	Reserved for future use

The active state of a bit shall be high (value 1). Bits that are reserved for future use shall be set to low (value 0).

The definition of the anomalies and defects linked to each of the indicator bits can be found in 10.5.4. The LoM-bit shall signal a loss of margin at the far end. It shall become high once loss of margin is detected and shall remain high as long as this condition exists.

8.5.5.4 Network Timing Reference (NTR)

Isochronous services require the same timing reference at transmit and receive sides in higher layers of the protocol stack. To support the transmission of this timing signal, the VDSL system shall transport an 8-kHz timing marker.

For applications that require NTR, NTR shall be transported in the following way:

The VTU-O shall derive a local 8-kHz timing reference (LTR), by dividing its sample clock with the appropriate number. For a VDSL system using $N_{sc} = 2^{n+8}$ tones, the sampling frequency could for instance be $2\, N_{sc}\, \Delta f$ and the dividing factor would then be $69 \times 2^{n+2}$.

The VTU-O shall estimate the change in phase offset between the NTR and the LTR from the previous superframe to the present. This value shall be expressed in cycles of a clock running at

frequency 2 N_{sc} Δf and shall be transported in the NTR overhead byte (see Table 8-3) as a 2's-complement number.

A positive value of the change in phase offset shall indicate that the LTR has a higher frequency than the NTR. A negative value of the change in phase offset shall indicate that the LTR has a lower frequency than the NTR.

The LTR, being proportional to Δf, has a maximum frequency variation of 50 ppm (see 9.2.1.1). The NTR has a maximum variation of 32 ppm. The maximum difference is therefore 82 ppm. This would result in a maximal phase offset of 0.205 µs per superframe. This corresponds to about $0.45.2^n$ samples. For the largest value of n (n = 4), this corresponds to somewhat more than 7 samples (in the positive or negative direction). One byte of information should therefore be sufficient.

8.5.6 Convergence of fast and interleaved buffer

Data from the interleaved and (optional) fast buffer shall be combined so that in each frame there shall first be a segment of fast data followed by a segment of interleaved data. Figure 8-4 illustrates this process.

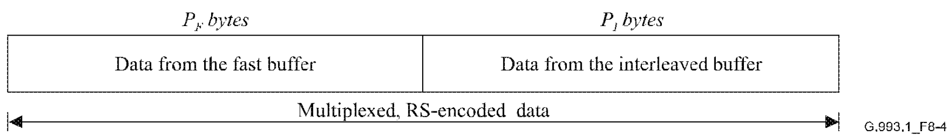

Figure 8-4/G.993.1 – Convergence of the fast and interleaved data into one frame

The total number of RS-encoded bytes per frame, P_{total}, is given by:

$$P_{total} = P_I + P_F$$

where P_I and P_F are the number of RS-encoded bytes from the interleaved and fast paths.

9 PMD sublayer

9.1 PMD functional model

The functional model of the PMD sublayer is presented in Figure 9-1.

EXHIBIT D

FILE HISTORY

60/549,804

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NEW PROVISIONAL APPLICATION TRANSMITTAL LETTER

Sir:

Transmitted herewith for filing is the Provisional Patent Application of Inventor(s):

	Marcos C. Tzannes
Residence:	121 LaEspiral Orinda, CA 94563
Citizenship	U.S.A.
Post Office Address:	Same as above

PATENTS
T3653-8993PV01

For: On-Line Impulse Noise Protection (INP) Adaptation

Enclosed are the following papers required to obtain a filing date under 37 C.F.R. §1.53(c):

__ Sheets of Informal Drawings
10 Pages of Specification, Drawings & Tables
__ Claims
1 Appendices

The following papers, if indicated by an ☒, are also enclosed:

☐ A Declaration and Power of Attorney
☐ An Assignment of the invention
☐ An Information-Disclosure Statement, Form PTO-1449 and a copy of each cited reference
☐ A Small-Entity Declaration
☒ A Certificate of Express Mailing, Express Mail Label No. ET 104264368 US

Basic Fee: $160

☒ A check in the amount of $160 is enclosed to cover the Filing Fee.

Please address all communications and telephone calls to the undersigned.

Respectfully submitted,

Gail Leonick

Gail Leonick
Aware, Inc.
40 Middlesex Turnpike
Bedford, Mass. 01730

Summary of Appendices

1. ITU – Telecommunication Standardization Sector
   Study Group 15
   Source: SBC, BellSouth, BT
   Title: Proposed revisions to G.992.4 and G.992.3&5 regarding impulse noise protection.

PATENT
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UNITED STATES PROVISIONAL PATENT APPLICATION

Of

Marcos C. Tzannes

for a

On-Line Impulse Noise Protection (INP) Adaptation

On-Line Impulse Noise Protection (INP) Adaptation

By Marcos Tzannes

Background

Communication systems often operate in environments with Impulse Noise. Impulse Noise is a short-term burst of noise that is higher than the normal noise that typically exists in the communication channel. For example, DSL systems operate on telephone lines and experience Impulse Noise from many external sources including telephones, AM radio, HAM radio, other DSL services on the same line or in the same bundle, other equipment in the home, etc. It is standard practice for communications systems to use Interleaving in combination with Forward Error Correction (FEC) to correct the errors caused by the Impulse Noise.

One of the problems with these communications systems is that they use traditional Signal to Noise Ratio (SNR) measurement techniques to determine the SNR of the channel. These traditional techniques assume that the noise is stationary and does not contain non-stationary components such as Impulse Noise. The most common method for measuring SNR is to calculate the mean-squared error of the received signal based on a known transmitted signal (see the ADSL series of ITU G.992.x standards and the VDSL series of ITU G.993.x. standards). These traditional methods for measuring SNR do not correctly measure the impact of Impulse Noise and do not have the capability to determine how the system should be configured to handle this Impulse Noise.

As an example, ITU-T Contributions SS-049 (attached) has proposed that there is a need in ADSL and VDSL systems to provide robust, error free performance in the presence of high real-world impulse noise. SS-049 proposes that the standard *Impulse Noise Protection (INP)* values are extended to values of 4, 8, 16 and 32 in order handle these high levels of impulse noise. INP is defined in the ADSL2 Standard G.992.3, which is incorporated herein by reference in its entirety, as the number of Impulse Noise corrupted DMT symbols that can be corrected by the FEC and Interleaving configuration. Specifically G.992.3 defines the following variables:

INP=1/2*(S*D)*R/N

S=8*N/L

Latency (or delay)=S*D/4

Line Rate (in kbps)=L*4

where N is the Codeword Size in bytes, R is the number of parity (or redundancy) bytes in a codeword, D is the interleaver depth in number of codewords and L is the number of bits in a DMT symbol. If K is the number of information bytes in a codeword then N=K+R and the User Data Rate is approximately equal to L*4*K/N.

SS-049 proposed that the higher INP values are achieved by increasing the amount of FEC redundancy while keeping the same system latency (and same Interleaver Memory) at the expense of user data rate or excess margin. Since, on phone lines without excess margin, there is clearly a tradeoff between high INP values and user data rate, it is best to

try to maximize the user data rate by finding the minimum INP value that will provide adequate impulse noise protection. The approach that is currently proposed in SS-049 involves the steps of:

1. The operator (or service provider) configures the ADSL connection with a specific INP value
2. The ADSL connection is initialized and transceivers enter into while in steady-state data transmission (also known as Showtime for DSL systems)
3. If the connection is stable (error-free) then the ADSL service is acceptable and the process ends. If there are bit errors then the process is repeated by going back to step #1.

An exemplary problem with process is that it is time consuming and can result in sub-optimum user data rates. To illustrate this lets take the following examples:

Example #1 (Time consuming process): Assume that for a particular DSL connection there is high impulse noise and the required INP is 8. As a result, if the service provider uses a first INP configuration of 2, the DSL connection will not be error free. Therefore the service provider needs to configure a higher INP value and reinitialize the connection. If a value of 4 is used as the second INP value, it still will not provide adequate impulse noise protection and the bit errors will occur. So the service provider will once again need to configure a higher INP value and so on, until the correct value of 8 is configured. Obviously the connection needs to be re-initialized every time there is a new INP configuration and this trial and error technique is very time consuming.

Example #2 (Sub-optimum User Data Rates): Assume that for a particular DSL connection there is high impulse noise and the required INP is 4. As a result, if the service provider uses a first INP configuration of 2, the DSL connection will not be error free. Therefore the service provider needs to configure a higher INP value and reinitialize the connection. However, in order to save time and not go through a number of initializations as in Example #1, the service provider simply configures the system to the maximum INP value of 32. Obviously there will be no errors with INP=32 since this connection only needed an INP value of 4. But as a result the user data can be greatly degraded since the additional FEC redundancy will be 3 times higher than what is actually needed. For example, if the an INP of 4 requires 10% FEC redundancy, an INP of 32 requires 40% FEC redundancy which results in a 30% decrease user data rate.

Overview of an Exemplary Embodiment of the Invention

An exemplary embodiment of this invention broadly describes a new method and system with the capability to determine the impact of Impulse Noise on a communication system and the capability to determine how the system should be configured to handle this Impulse Noise.

According an exemplary embodiment of this invention, the impact of impulse noise is determined by transmitting and receiving using a plurality of different Forward Error Coding (FEC) and Interleaving Parameter settings. For each FEC and Interleaving Parameter (FIP) setting the received signal quality is determined by, for example,

detecting if there are bit errors after the receiver performs the FEC decoding and deinterleaving. Based on this, the appropriate FIP setting is selected and used for transmission and reception.

As described above, DSL systems (such as the one defined in ADSL G.992.x or VDSL G.993.x) use FIP parameters denoted as N, K, R and D. For these systems an FIP setting can be characterized by the set of parameters (N, K, R, D). Using these parameters, the Burst Error Correction Capability (BECC) in bytes can be simply calculated as BECC=D*R/2 bytes where BECC is defined as the number of consecutive byte errors that can be corrected by the receiver. Note that if the receiver uses more intelligent decoding schemes (e.g. erasure detection) it is possible to correct even more than D*R/2 bytes. It also follows from above that INP=BECC/L. For example a setting of (N=255,K=239,R=16,D=64) would correspond to N=255 bytes, K=239 bytes, R=16 bytes, D=64 codewords and in this case the receiver can correct BECC=D*R/2=64*16/2=512 bytes. If L=256*8 bits, this corresponds to a line rate of approximately 4*(256*8)=8.192 Mbps for standard ADSL and VDSL systems and INP=512/256=2.

According exemplary embodiments of this invention, a plurality of FIP settings are used for transmission and reception. In an exemplary embodiment, the system transitions from one FIP setting to another FIP setting is completed without going through a new initialization procedure such as the one specified in traditional xDSL systems. For example, an xDSL system that implements the method of this invention could start using a FIP setting of (N=255,K=247,R=8,D=64) and then change to a setting of (N=255,K=239,R=16,D=64) without completing a new initialization procedure. Note that the first FIP setting has a BECC=256 bytes and the second setting has a BECC=512 bytes, which means, for example, that the second setting can correct an impulse that causes twice the as many byte errors as the first FIP setting. On the other hand, the first FIP setting has less FEC parity (overhead), which results in a higher information (net) data rate for the user. This can also be seen by the fact that K, the number of information bytes per codeword, is higher for the first FIP setting. For each of the FIP settings the receiver detects whether there are bit errors after the decoding and deinterleaving process. This detection can be done by, for example, performing a Cyclic Redundancy Check (CRC) after the decoding/deinterleaving process as is defined in ITU standard G.992.x. In general, a CRC is a well-known method for detecting bit errors. Since Impulse Noise occurs at random times, the system operates using a particular FIP setting for a period of time that is sufficient to encounter the Impulse Noise. In the simple example directly above, only the K and R values were modified. This invention is not limited in any way in this manner but in fact any combination of FIP parameters can be modified in the process.

The process of determining the impact of impulse noise by transmitting and receiving using a plurality of FIP settings can be done while in steady-state transmission (also known as Showtime for ADSL systems) when user information bits are being transmitted. Alternatively the process of determining the impact of impulse noise by transmitting and receiving using a plurality of FIP setting can be done during a special

Impulse Noise Training period during which the system is not actually transmitting user data. For example, in this special Impulse Noise Training period the system could transmit idle ATM cells or HDLC flags or a predefined pseudorandom bit sequence.

Throughout this invention the term “transmitter” has the same meaning as the term “transmitting modem”. Throughout this invention the term “receiver” has the same meaning as the term “receiving modem”.

Detailed Description: On-Line INP Adaptation

According to an exemplary embodiment of this invention, the communication system adapts the INP setting on-line by operating using a series of different FIP settings. For each FIP setting the system determines if the appropriate amount of impulse noise protection (INP) is being provided. Based on these determinations the system selects a particular FIP setting for regular operation.

This On-Line INP adaptation procedure comprises the following exemplary steps:

1. The DSL system completes regular initialization and starts transmitting and receiving using a first FIP setting. For DSL systems this FIP setting is selected by the receiver and is based on the min/max data rate, max latency and min INP values as configured by the service provider via the CO-MIB (see G.992.3). Proceed to step 2.
2. The system operates using this FIP setting for a period of time T1. Proceed to Step 3.
3. The receiver detects if bit errors have occurred using this FIP setting for Decoding and Deinterleaving during the period of time T1. For example the receiver could use a CRC to detect bit errors. If there are no bit errors then the current INP setting is adequate and there is no need to change the INP setting on-line and the On-Line INP adaptation procedure ends. On the other hand, if there are CRC errors proceed to step 4.
4. Since there are bit errors, the service provider or user may choose to request an on-line increase of the INP setting (for example double the INP setting). Based on this request the INP setting would be increased by on-line modification of the FIP parameters. Go to step 5.
5. In order initiate the change in the INP setting, the ATU-C may send a message to the ATU-R that specifies the new INP value. Go to step 6.
6. The receiver sends a message to the transmitter indicating the new FIP setting that satisfies the new INP setting. Go to step 7.
7. The transmitter and receiver transition to the new INP setting by starting to use the new FIP parameters for transmission and reception, respectively, at a synchronized point in time. This synchronization could be done in a number of different ways (see section below). Go to step 8.
8. The system operates using this new FIP setting for a period of time T2. Proceed to Step 9.
9. The receiver detects if bit errors have occurred using this FIP setting for Decoding and Deinterleaving during the period of time T2. If there are no bit errors detected by the receiver during the period of time T2 then the new INP setting is adequate and there is no need to change the INP setting on-line and the On-Line INP adaptation procedure ends. On the other hand, if there are bit errors go back to step 4 and repeat the process to select a new INP setting.

The On-line INP adaptation process could be repeated as many times as desired until an INP setting is selected that provides the required impulse noise immunity.

There are several important points regarding these exemplary embodiments:

- The transition between different FIP settings is done without reinitializing the transceivers using a lengthy initialization procedure such as is used in ADSL and VDSL systems.
- The transition between FIP settings can be synchronized between the transmitter and the receiver so that the receiver can determine when to use start FEC decoding using the new FIP settings for K and R. The transition could be synchronized using a number of methods. The section below describes some of these methods.
- Alternatively the transition could be done without synchronization in which case the receiver would have to determine when the new FIP setting are used by some other means, e.g. by FEC decoding using both FIP settings and determining which one is being used by calculating whether the codeword is correct with one setting or the other.
- These steps can be performed during regular steady state transmission (also known as SHOWTIME in ADSL) using actual user data or, for example, idle ATM cells.
- Alternatively these steps can be performed during a special Impulse Noise Training period during which the system is not actually transmitting user data. For example, in this special Impulse Noise Training period the system could transmit predefined pseudorandom bit stream or, for example, idle ATM cells or HDLC flags.
- The length of the time periods T1, T2, etc can be controlled by the receiver. The receiver could control the length of these time periods by for example sending a message that specifies how long the transmitter should transmit using a particular FIP setting. This length can be defined, for example, in terms of the number of DMT symbols or the number FEC codewords. For example, the message could indicate that 200 DMT symbols should be sent for all FIP settings or 300 FEC codewords should be sent for all FIP settings. Alternately the message could indicate a different number of DMT symbols or FEC codewords for each FIP setting.
- The length of the time periods T1, T2, etc can be controlled by the transmitter. The transmitter could control the length of these time periods by for example sending a message to the receiver that specifies how long the transmitter will transmit using a particular FIP setting. This length can be defined in terms of the number of DMT symbols or the number of FEC codewords. For example, the message could indicate that 200 DMT symbols will be sent for all FIP settings or 300 FEC codewords will be sent for all FIP settings. Alternately the message could indicate a different number of DMT symbols or FEC codewords for each FIP setting.
- The length of the time periods T1, T2 and T3, etc can be also controlled by the service provide or the user. For example the service provider could configure through the CO-MIB that a minimum of time of X=10 seconds be used for testing

each FIP setting. The service provider could use knowledge of the nature of the impulse noise (e.g. how often the impulse noise occurs) to determine these times.

- In Step #4 of the INP Adaptation process the service provider chooses to request a new INP setting. Alternatively the receiver can make this choice. In this case, the receiver could continually adapt the INP value as described above until the are no bit errors due to impulse noise.

Synchronizing the modification of the FEC and interleaving parameters

As stated in exemplary embodiments above, the receiver and transmitter can synchronize the modification of the FEC and interleaving parameters so that they both start using them at the same instant in time. There are several ways to do this.

Synchronizing using FEC codeword counters: In this example, the receiver and transmitter synchronize the change by counting FEC codewords from the beginning of Showtime and the transition occurs on a specific FEC codeword counter value that is known by both the transmitter and the receiver. Prior to this the receiver or the transmitter will send a message to the other side indicating the FEC codeword count value on which the FIP parameters will be modified. For example the exemplary process is as follows:

1. The transmitting modem enters Showtime and starts a counter that counts the number of transmitted FEC codewords, where the first codeword transmitted has a count value of 0, the second transmitted codeword has count value of 1 and so on. From practical implementations this counter will probably have a finite length, for example 0 to 1023 (10 bits) so that when the value of 1023 is reached, on the next FEC codeword the counter starts at the value of 0 again.
2. Likewise, the receiving modem enters Showtime and starts a counter that counts the number of received FEC codewords, where the first codeword received has a count value of 0, the second received codeword has count value of 1 and so on. From practical implementations this counter will probably have a finite length, for example 0 to 1023 (10 bits) so that when the value of 1023 is reached, on the next FEC codeword the counter starts at the value of 0 again.
3. At some point in time it is determined that a new a FIP setting is needed due to the presence of Impulse Noise on the line. This determination can be done by the receiving modem, the transmitting modem, the user or service provide (operator).
4. The receiving modem sends a message to the transmitting modem indicating the new FIP setting to be used for transmission and reception. Alternatively, the transmitting modem sends a message to the receiving modem indicating the new FIP setting to be used for transmission and reception
5. The receiving modem sends a message to the transmitting modem indicating the FEC codeword counter value on which the new FIP settings are to be used for transmission and reception. Alternatively, the transmitting modem sends a message to the receiving modem indicating the FEC codeword counter value on

which the new FIP settings are to be used for transmission and reception. For example the message could indicate that when the codeword counter equals 501 the new FIP setting will be used for transmission and reception.

6. When the transmitter FEC codeword counter equals the value indicated in the message, the transmitting modem uses the new FIP settings for transmission.
7. Likewise, when the receiver FEC codeword counter equals the value indicated in the message, the receiving modem uses the new FIP settings for reception.

Synchronizing using a Flag Signal: In this example, the receiver and transmitter synchronize the change using a Flag or Marker signal (similar to the signal used in the ADSL2 G.992.3 OLR protocol.) This protocol may be more desirable that using an FEC codeword counter because it has more immunity to impulse noise. In this case the receiver and transmitter would start using the new FEC and Interleaving parameters on a predefined FEC codeword boundary following the SyncFlag. For example the process is as follows:

1. While transmitting using a first INP setting, it is determined that a new a FIP setting is needed due to the presence of Impulse Noise on the line. This determination can be done by the receiving modem, the transmitting modem, the user or service provide (operator).
2. The receiving modem sends a message to the transmitting modem indicating the new FIP setting to be used for transmission and reception. Alternatively, the transmitting modem sends a message to the receiving modem indicating the new FIP setting to be used for transmission and reception
3. The transmitting modem sends a Flag or Marker Signal to indicate that the new FIP setting are to be used on a predetermined number of DMT symbols following the transmission of the Flag or Marker Signal. For example the Flag signal could be an inverted Sync Symbol, or SyncFlag, as is used in the ADSL2 G.992.3 OLR protocol.
4. The transmitting modem start using the new FIP settings for transmission on the predetermined number of DMT symbols following the transmission of the Flag or Marker Signal.
5. Likewise, the receiving modem start using the new FIP settings for reception on the predetermined number of DMT symbols following the reception of the Flag or Marker Signal.

Example #1 of On-Line INP adaptation FIP Setting

This section describes an example of FIP settings for On-Line INP adaptation for DSL. In this example only the number of information bytes in a codeword (K) and the number of parity bytes in a codeword (R) are updated on line. The Codeword Size (N) and

Interleaver Depth (D) are not changed. This means that the latency (and interleaver memory size) and the line rate are not modified on-line. Since N=K+R this places restrictions on the allowed values for K and R.

1st Setting – {Approximate User Data Rate= 3.968 Mbps, Line Rate=4.096 Mbps, N=128, K=124, R=4, S=1, D=64, Latency=16 msec, INP=1}

2nd Setting - {Approximate User Data Rate= 3.840 Mbps, Line Rate=4.096 Mbps, N=128, K=120, R=8, S=1, D=64, Latency=16 msec, INP=2}

3rd Setting - {Approximate User Data Rate= 3.584 Mbps, Line Rate=4.096 Mbps, N=128, K=112, R=16, S=1, D=64, Latency=16 msec, INP=4}

The On-line INP adaptation process is restricted to only modify the number of information bytes in a codeword (K) and the number of parity bytes in a codeword (R). The FEC Codeword Size (N=K+R) and Interleaver Depth (D) are not changed. This means that the latency (or interleaver memory size) and the line rate are not modified on-line. However the user data rate will change during the process since K is being modified. Also since the line rate and the FEC codeword size are not modified, the S value does not change in the process. It is important to note that with these constraints, the On-line INP adaptation process can be done in a seamless manner (no bit errors and service interruption). This means provided that the modification if the FIP setting is restricted to K and R, the transition between FIP settings can be done in a seamless manner. This is the case because if the codeword size N and the interleaver depth D are not modified, the transition can happen without the problem of "interleaving memory flushing." Interleaver memory flushing is a well-known problem in which errors occur because interleaver and deinterleaver memory locations are overwritten due to on-line changes in the codeword size (N) and or interleaver depth (D).

Example #2 of On-Line INP adaptation FIP Setting

This section describes an example of FIP settings for On-Line INP adaptation for DSL. In this example only the Codeword Size (N) and the number of parity bytes in a codeword (R) are updated on line. The number of information bytes in a codeword (K) and Interleaver Depth (D) are not changed and therefore the user data rate does not change. This means that the latency (and interleaver memory size) and the line rate are modified on-line. Since N=K+R this places restrictions on the allowed values for K and R.

1st Setting – {Approximate User Data Rate= 3.968Mbps, Line Rate=4.096 Mbps, N=128, K=124, R=4, S=1, D=64, Latency=16 msec, INP=1}

2nd Setting - {Approximate User Data Rate= 3.968 Mbps, Line Rate=4.224 Mbps, N=132, K=124, R=8, S=1, D=64, Latency=16 msec, INP=2}

3rd Setting - {Approximate User Data Rate= 3.968 Mbps, Line Rate=4.480 N=128, K=124, R=16, S=1, D=64, Latency=16 msec, INP=4}

In this example the Line Rate is modified on-line. For this reason it is necessary to also complete a Rate Adaptation process in order to complete this On-Line INP adaptation. A method for Seamless Rate Adaptation is described in US Patent 6,498,808 which is incorporated herein in its entirety.

While these examples restrict the changes to a subset of the FIP parameters, they can obviously be extended to cover any combination of the FIP parameters (N, K, R and D). For example, the value of D could also be modified in addition to the values of K, R and N. This could result in a change in the required interleaver memory and latency. In order to keep the memory and latency constant it is necessary to change the codeword size (N) accordingly when changing the interleaver depth (D). For example, if the interleaver depths is changed from D=64 to D=128, the Codeword size would have to be decreased by a factor of 2 so that overall latency is constant.

The above-described communication system can be implemented on wired or wireless telecommunications devices, such a modem, a multicarrier modem, a DSL modem, an ADSL modem, an XDSL modem, a VDSL modem, a multicarrier transceiver, a wired or a wireless wide/local area network system, or the like, or on a separate programmed general purpose computer having a communications device. Additionally, the systems, methods and protocols of this invention can be implemented on a special purpose computer, a programmed microprocessor or microcontroller and peripheral integrated circuit element(s), an ASIC or other integrated circuit, a digital signal processor, a hard-wired electronic or logic circuit such as discrete element circuit, a programmable logic device such as PLD, PLA, FPGA, PAL, modem, transmitter/receiver, or the like. In general, any device capable of implementing a state machine that is in turn capable of implementing the methodology illustrated herein can be used to implement the various communication methods, protocols and techniques according to this invention.

Furthermore, the disclosed methods may be readily implemented in software using object or object-oriented software development environments that provide portable source code that can be used on a variety of computer or workstation platforms. Alternatively, the disclosed system may be implemented partially or fully in hardware using standard logic circuits or VLSI design. Whether software or hardware is used to implement the systems in accordance with this invention is dependent on the speed and/or efficiency requirements of the system, the particular function, and the particular software or hardware systems or microprocessor or microcomputer systems being utilized. The communication systems, methods and protocols illustrated herein however can be readily implemented in hardware and/or software using any known or later developed systems or structures, devices and/or software by those of ordinary skill in the applicable art from the functional description provided herein and with a general basic knowledge of the computer and telecommunications arts.

Moreover, the disclosed methods may be readily implemented in software executed on programmed general purpose computer, a special purpose computer, a microprocessor, or the like. In these instances, the systems and methods of this invention can be implemented as program embedded on personal computer such as JAVA® or CGI script, as a resource residing on a server or graphics workstation, as a routine embedded in a dedicated communication system, or the like. The system can also be implemented by physically incorporating the system and method into a software and/or hardware system, such as the hardware and software systems of a communications transceiver.

ITU - Telecommunication Standardization Sector **SS-049**

STUDY GROUP **15**

Singapore, January 19-24 2004

Question: 4/15

SOURCE*: SBC, BellSouth, BT

TITLE: Proposed revisions to G.992.4 and G.992.3&5 regarding impulse noise protection

ABSTRACT

SBC field experience has shown that some ADSL lines fail to provide reliable service due to high level intermittent noise that has frequent re-occurrence. This differs from the classic concept of short duration and infrequent impulse noise. SBC field experiments have determined that many trouble cases due to intermittent noise can be effectively resolved by setting the ADSL coding parameters to provide a high level of coding redundancy. The necessary coding can reach 50% redundancy in some cases, while having an additional latency of 10 ms or less. The necessary coding redundancy exceeds the current specifications in the ITU ADSL Recommendations G.992.3 and G.992.5. It is proposed that the "minimum impulse noise protection (INP)" parameter in these Recommendations be revised to allow values up to 32 symbols and to change its name to "impulse strength indication".

1. Discussion

G.997.1 defines a parameter in Section 7.3.2.2 called "minimum impulse noise protection (INP)" that is measured in symbol periods and currently allowed to take one of the 4 values 0, .5, 1, and 2 symbols (corresponding to 0 125, 250, and 500 microseconds of a theoretical impulse). Section 7.3.2.3 also defines a maximum delay. The parameter is passed in G.994.1 and further described in Annex K of G.992.3. Both of these parameters are active in G.992.3/4/5 and actually exchanged during the "handshake" portion of training and so specified from modem to modem via the NPAR(3) octet of G.994.1. It is proposed that this section and parameter be renamed "impulse strength indication" and for the downstream direction takes on the additional values 4, 8, 16 and if possible values between 16 and 32. Note that there NPAR(3) uses bits 3

* Tom Starr — Tel: +1 (847) 248-5467
SBC — Email: tom.starr@ameritech.com

John Cioffi — Tel: +1-925-823-2910
SBC — Email: jc7513@sbc.com

Gary Tennyson — Tel: +1-205-985-6087
BellSouth — Email: gary.tennyson@bellsouth.com

Kevin Foster — Tel: +44-1473-642-986
BT — Email: kevin.t.foster@bt.com

and 4 today, allowing only the present 4 options. However, bit 5 is fortunately reserved by the ITU and so could be additionally allocated to INP also, thus allowing the desired 8 options proposed here. The ratio of this INP parameter to interleaving depth will then determine the codeword block size. The codeword block size will be supplied by the receiving modem back to the transmitter in a message known as C/R-PAREMS of the exchange of G.992.3 and G.992.5 ADSL modems. The specific parameters that will reflect the additional intermittent-noise protection will be the $B_{p,n}$ and $R_{p,n}$ that are determined by the receiver and passed to the transmitter in the usual manner.

A loading-algorithm engineer could use the following formulas to convert the supplied maximum delay and now renamed "impulse strength indication INP" into the usual FEC parameters (where $t = R$, the number of parity bytes chosen by the receiver, if erasures are used and $t = R/2$ if no erasures are used and rate includes information and synch/control bits)

$$D' = \text{depth in bytes} = \frac{(INP/4)\times\left(\frac{\text{rate in kbps}}{8}\right)}{\text{t}} \qquad (1)$$

$$N = \frac{(\text{delay in ms})\times\left(\frac{\text{rate in kbps}}{8}\right)}{(D'-1)} + 1 \approx \frac{\text{delay}}{\text{burst length}}\cdot t \qquad (2)$$

Then the parameters M, T, L, etc can be computed and S inferred for all latency paths as well as the values of B for each frame bearer and latency path. Also once N is known, then $D = ND'$. Even approximate adherence to the above mathematical rules should solve the problem. Service providers may well train the modem several times, each time increasing the INP value until zero or small numbers of code violations and/or error seconds are observed over a time period determined by the service provider.

We note here that even with the choice of smallest delay (4 ms) and largest burst length in the existing G.992.3 provides the relation

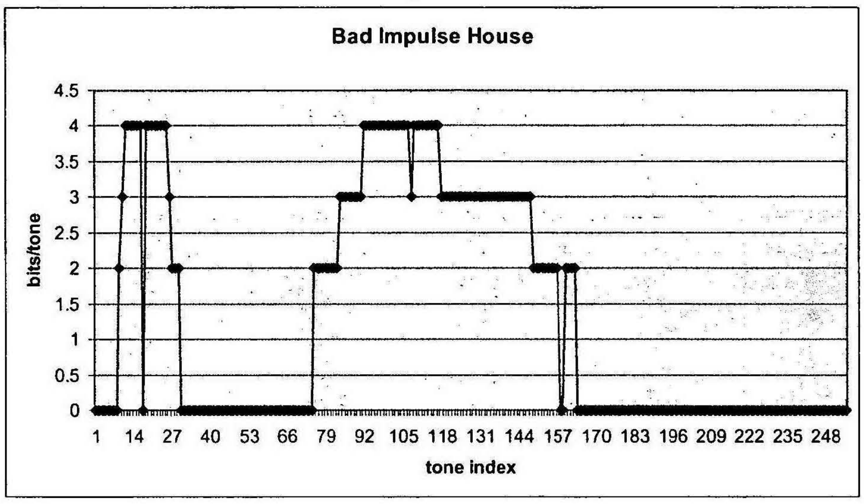
$$N \approx 8 \cdot t$$

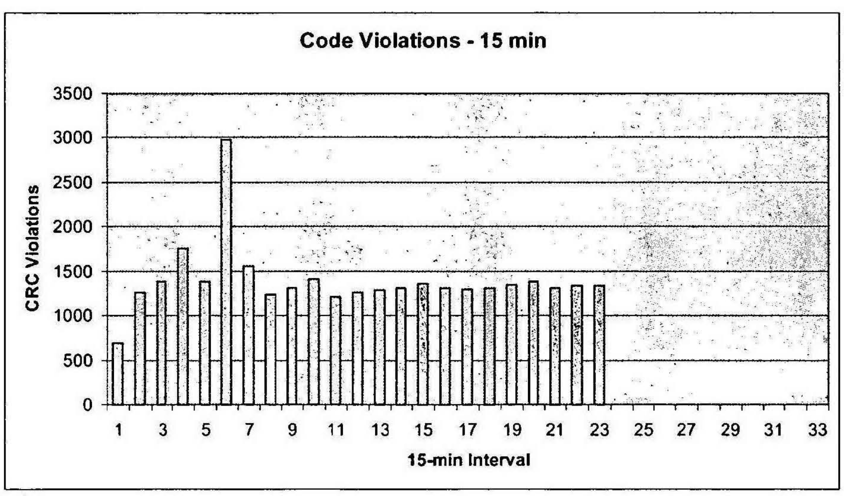
in equation (2) above. Since t is usually 16 in well-designed systems that have maximum parity then, the smallest codeword length is N=128, much larger than the values such as 32, 48, 64, and 80 that are effective for chronic lines with intermittent noise. The addition of the new values of INP=4,8,16, or 32 allows this expression to allow very small N values essentially providing service providers with the ability they need to correct a line undergoing difficult intermittent noise (even if at low data rate). The value of INP=8 allows codeword lengths as low as 32, and the higher values allow very low data rates on severely chronic lines.

An INP setting of 4 or larger will imply that impulse protection is more important than maximum data rate and so the equipment will connect at the highest data rate that maintains the impulse protection level indicated regardless of the net data rate. Service providers are expected to use the INP setting of 4 only for lines with severe noise, where the lower data rate is a better choice than not having reliable service.

2. Sample Example

The following example was taken from chronic lines reported by the SBC Broadband Tools. The customer operates in 768 kbps interleave profile, but has 28 dB margin downstream (and a maximum attainable data rate of over 4.5 Mbps reported). While the margin is high, the bit distribution clearly indicates that an intermittent noise is affecting frequencies below 400 kHz so often that the modem will not load into those frequencies. Even as such with the high margin indicated, the error seconds just below indicate a severe problem on the line. This line will operate with no errors at 1.5 Mbps if the for the combination $N=24$ and $R=16$. Such a setting is not a current possible selection by service providers if DSLAM profiles or MIBS do not have this choice (yet). The customer probably switched to cable-modem by the time this note was read, but easily could have been better served by DSL at a higher rate with no errors in ADSL1.





In specially investigating about 50 customers, the authors found 2 other lines that also could only be fixed (and would run at least 1.5 Mbps with zero errors instead of being out of service or chronic) by lowering the N parameter (or equivalently choosing a larger value for the "impulse strength" in ADSL2).

3. Proposal

1- Agree to an amendment for corrigendum to ITU G.992.3, G.992.5 G.994.1, G.997.1 to change the name of the parameter "minimum impulse noise protection (INP)" to "impulse strength indication".

2- In the above Recommendations, revise the definition of this parameter to have the additional values 4, 8, 16 and if possible values between 16 and 32.

3- In ITU G.994.1 NPAR(3) for this parameter, define bit 5 to allow for the additional defined values.

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out of the incomplete transaction. This allows you to maintain your database's integrity. You may, however, lose the single transaction you were working on when your network got sick.

Transaction Processing A processing method in which transactions are executed immediately when they are received by the system, rather than at some later time as in batch-processing systems. Airline reservation databases and automatic teller machines are examples of transaction-processing systems.

Transactional Integrity A term that describes how your computing/telecom system handles making sure that the transaction you just made is solid and clean and that the next time you want to get to the results of the transaction you can. "Transactional integrity" becomes critical when you're storing bits and pieces of your transactions on different media, in different places. For example, you might want to store your data on a magnetic hard drive and your associated images on a separate optical drive.

Transborder Data Flow TDF. Transborder data flows are movements of machine-readable data across international boundaries. TDF legislation began in the 1970s and has been put into effect by many countries in an attempt to protect personal privacy of citizens. This term has particular meaning as it relates to electronic commerce or EDI and is becoming more and more relevant with the use of the Internet as a means to conduct global business.

Transceiver 1. Any device that transmits and receives. In sending and receiving information, it often provides data packet collision detection as well.

2. In IEEE 802.3 networks, the attachment hardware connecting the controller interface to the transmission cable. The transceiver contains the carrier-sense logic, the transmit/receive logic, and the collision-detect logic.

3. A device to connect workstations to standard thick Ethernet-style (IEEE 802.3).

Transceiver Cable In local area networks, a cable that connects a network device such as a computer to a physical medium such as an Ethernet network. A transceiver cable is also called drop cable because it runs from a network node to a transceiver (a transmit / receiver) attached to the trunk cable. See Transceiver.

Transcoder A device that combines two 1.544 megabit per second bit streams into a single 1.544 megabit per second bit stream to enable transmission of 44 or 48 voice conversations over a DS-1 medium.

Transcoding A procedure for modifying a stream of data carried so that it may be carried via a different type of network. For example, transcoding allows H.320 video encoding, carried via circuit switched TDM systems to be converted to H.323 so that it can connect with and be transmitted across packet switched ethernet LAN.

Transcriptionist A person who listens to a tape recording and types the words he hears. The word, transcriptionist, derives from the verb to transcribe. The most common employment of transcribers is in the medical industry, where busy doctors talk into tape recorders telling good and bad news of their patients. And even busier transcriptionists type those words into the patient's medical records, or whatever.

Transducer A device which converts one form of energy into another. The diaphragm in the telephone receiver and the carbon microphone in the transmitter are transducers. They change variations in sound pressure (your voice) to variations in electricity, and vice versa. Another transducer is the interface between a computer, which produces electron-based signals, and a fiber-optic transmission medium, which handles photon-based signals.

Transfer A telephone system feature which provides the ability to move a call from one extension to another. It is probably the most commonly used and misused feature on a PBX. Before you buy a PBX, check out how easy it is to transfer a call. If you have a single line phone, you should simply hit the touch hook, hear a dial tone and then dial the chosen extension number and hang up. This sounds easy in principle, but many people find it difficult since they associate the touch hook with hanging up the phone. Some companies have gotten around this by putting a "hook flash" button on the phone itself. Such a button is like having an autodial button which just makes the exact short tone you make when you quickly hit the hook flash button. An even better solution is an electronic phone with a button specially marked "transfer," or a button next to a screen which lights up "transfer." Failing to efficiently transfer a call is the easiest way to give your customers the wrong impression of your firm. Think of how many times have you called a company only to be told it wasn't the fellow's job and he will transfer the call, but "If we get cut off, please call Joe back on extension 2358." There are typically four types of Transfer: Transfer using Hold, Transfer using Conference, and Transfer with and without Announcement.

Transfer Callback A phone system feature. After a specified number of rings, an unanswered transferred call will return to the telephone which originally made the transfer.

Transfer Delay A characteristic of system performance that expresses the time delay in processing information through a data transmission system.

Transfer Impedance A measure of shield effectiveness.

Transfer Mode A fundamental element of a communications protocol, transfer mode refers to the functioning arrangement between transmitting and receiving devices across a network. There are two basic transfer modes: connection-oriented and connectionless. Connection-oriented network protocols require that a call be set up before the data transmission begins, and that the call subsequently be torn down. Further, all data are considered to be part of a data stream. Examples of connection-oriented protocols include analog circuit-switched voice and data, ISDN, X.25 and ATM.

Connectionless protocols, on the other hand, do not depend on such a process. Rather, the transmitting device gains access to the transmission medium and begins to transmit data address to the receiver, without setting up a logical connection across the physical network. LANs (e.g. Ethernet and Token Ring) make use of connectionless protocols, as does SMDS, which actually is an extension of the LAN concept across a MAN (Metropolitan Area Network). For more detail, see Connection Oriented and Connectionless Mode Transmission.

Transfer Protocols Protocols are all of the packaging" that surround actual user data to tell the network devices where to send the data, who it comes from, and how to tell if it arrived. Transfer protocols are designed for the efficient moving of larger chunks of user data.

Transfer Rate The speed of data transfer — in bits, bytes or characters per second — between devices.

Transfer Switch Usually a switch which reverses two input-output combinations.

Transfer Time A power backup term. Transfer time can refer to either the speed to which an off-line UPS transfers from utility power to battery power, or to the speed with which an on-line UPS switches from the inverter to utility power in the event of an inverter failure. In either case the time involved must be shorter than the length of time that the computer's switching power supply has enough energy to maintain adequate output voltage. this hold-up time may range from eight to 16 milliseconds, depending on the point in the power supply's recharging cycle that the power outage occurs, and the amount of energy storage capacitance within the power supply. A transfer time of 4ms is most desirable , however, it should be noted that an oversensitive unit may make unnecessary power transfers.

Transformer Transformers are devices that change electrical current from one voltage to another. A step-up transformer increases the voltage and a step-down transformer decreases voltage. The power of an electric current must be conserved so just as voltage is increased, current is decreased. Transformers work by feeding an alternating current into a primary coil. The primary coil induces a magnetic field in a secondary coil which is connected to an energy using load. The difference between the number of coils in the primary coil versus the secondary coil determines whether the voltage will be stepped up or down. One reason for using a transformer is that commercial power is typically 120 or 240 volts while many phone systems (and other computer-type "things") work best on 48, 24 or lower voltage.

Transformer Exciting Network See TEN.

Transhybrid Loss The transmission loss between opposite ports of a hybrid network, that is between the two ports of the four-wire connection.

Transient Any high-speed, short duration increase or decrease impairment that is superimposed on a circuit. Transients can interrupt or halt data exchange on a network. See HIT.

Transient Mobile Unit A mobile unit communicating through a foreign base station.

Transistor The transistor was invented in 1947 by John Bardeen, Walter H. Brattain and William Shockley of Bell Laboratories. The first transistor comprised a paper clip, two slivers of gold, and a piece of germanium on a crystal plate. Here is an explanation of how a transistor works, taken from "Signals, The Science of Telecommunications" by John Pierce and Michael Noll:

"To understand how a transistor works, we must look at the laws of quantum mechanics. We commonly picture an atom as a positive nucleus surrounded by orbiting electrons ... Vacuum tubes rely on the ability of electrons to travel freely with any energy through a vacuum. Transistors rely on the free travel of electrons through crystalline solids called semiconductors ... Semiconductors (such as silicon or gallium arsenide) differ from pure con-

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2

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point must be reversed (*backed out*) and a DBMS must provide facilities to ensure this happens. When the processing of a transaction has been completed satisfactorily the changes to the database are *committed* – made permanent.

transaction file (movement file) A file, especially a *data file, containing transaction records, prior to the updating of a *master file. Transaction files are only used in *batch processing systems. Once updating has been carried out, the transaction file may be kept in order to permit subsequent recovery of the master file (*see* FILE RECOVERY) or for auditing purposes but is otherwise redundant.

transaction processing A method of organizing a data-processing system in which *transactions are processed to completion as they arise. A *transaction processing monitor* (*TP monitor*) is a software system that facilitates the handling of transactions in such circumstances. *Compare* BATCH PROCESSING.

transceiver A device that can both transmit and receive signals on a communication medium. Many communication devices, including *modems, *codecs, and *terminals, are transceivers.

transducer 1. (sensor) Any device that converts energy in the form of sound, light, pressure, etc., into an equivalent electrical signal, or vice versa. For example, a semiconductor laser converts electrical energy into light, and a piezoelectric device converts mechanical stress into electrical energy (and vice versa).
2. In formal language theory, any *automaton that produces output.

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transfer rate *See* DATA TRANSFER RATE.

transformation 1. *Another name for* function, used especially in geometry.
2. of programs. *See* PROGRAM TRANSFORMATION.
3. of statistics data. A change of scale used to improve the validity of statistical analyses. For data in which small values have smaller *variance than large values a logarithmic or square-root transformation is often recommended. For data in the form of proportions, a transformation from the scale (0,1) to an infinite scale is advisable before performing *analysis of variance or *regression analysis. Several transformations exist for proportions, such as the *logistic or log-odds-ratio that is used in the analysis of *generalized linear models. Appropriate transformations may be suggested by studying *residuals in a regression analysis.

transformational grammar A grammar that makes essential use of transformation rules to convert the *deep structures of sentences into their surface structures. *See also* GENERATIVE GRAMMAR.

transformational semantics *See* PROGRAM TRANSFORMATION.

transformation matrix An $m \times n$ matrix of numbers used to map vectors with n elements onto vectors with m elements.

transformation monoid *See* TRANSFORMATION SEMIGROUP.

transformation semigroup A *semigroup consisting of a collection C of transformations of a *set S into itself (*see* FUNCTION), the *dyadic operation $\circ$ being the *composition of functions; it is essential that the set C should be *closed with respect to composition, i.e. if c_1 and c_2 are in C then so is $c_1 \circ c_2$.
If the identity transformation (*see* IDENTITY FUNCTION) is included in the transformation semigroup, a *transformation monoid* results. Every monoid is isomorphic to a transformation monoid.

transform domain *See* FILTERING.

transient error An error that occurs once or at unpredictable intervals. *See also* ERROR RATE.

transistor A semiconductor device having, in general, three terminals that are attached to electrode regions within the device. Current flowing between two of these electrodes is made to vary in response to voltage or current variations imposed on the third electrode. The device is capable of current or voltage amplification depending on the particular circuit implementation employed. It can also be used as a switch by driving it between its maximum and minimum of current flow.
The transistor was invented in 1948 by Shockley, Brattain, and Bardeen at Bell Telephone Labs. As performance and manufacturing techniques improved, the tran-